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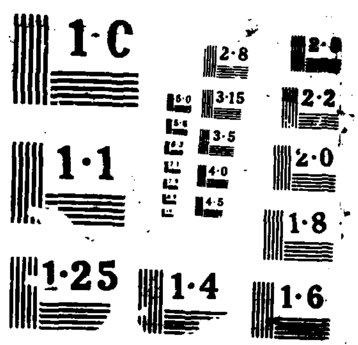

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SIMULATION AND ANALYSIS OF A LOCAL AREA NETWORK INTERCONNECTION BACKBONE FOR HQ, U.S. ARMY EUROPE AND THE SEVENTH ARMY

THESIS

Lawrence J. Shrader
Captain, USAF

AFIT/GCS/ENG/87D-25

DEPARTMENT OF THE AIR FORCE
AIR UNIVERSITY
AIR FORCE INSTITUTE OF TECHNOLOGY

Wright-Patterson Air Force Base, Ohio

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THESIS

Presented to the Faculty of the School of Engineering
of the Air Force Institute of Technology
Air University
In Partial Fulfillment of the
Requirements for the Degree of
Master of Science in Computer Systems

Lawrence J. Shrader, B.S.
Captain, USAF

December 1987

Approved for public release; distribution unlimited

Preface

The purpose of this thesis project was to analyze and model by simulation, the local area network interconnection backbone proposed for the information community at HQ, U.S. Army Europe and the Seventh Army located at Heidelberg, Germany.

The results of this project apply to performance of the proposed backbone. However, an effort was made to keep the code design modular so further development and expansion of the model can proceed as the sponsor sees fit.

I wish to thank Lieutenant Colonel Albert B. Garcia, my thesis advisor, for his assistance throughout the course of this project. I also thank Mr. Raymond Johnson, my sponsoring organization point of contact at the Information Systems Engineering Command - Europe, for his patience and help during my site visit.

Larry J. Shrader

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Abstract

This thesis project develops a simulation model of the proposed backbone that will interconnect approximately 16 existing local area networks of the information community at HQ, U. S. Army Europe and the Seventh Army in Heidelberg, Germany. The objective is to develop a computer simulation that determines throughput and delay characteristics of the backbone. Potential configuration problems that may limit the full utility of the system are also identified from results of the simulation.

An existing telephone switch at the Campbell Barracks in Heidelberg will be upgraded and interfaced, by way of a 9.6K bits/second link, to a packet switch to provide an Integrated Services Digital Network (ISDN) environment for the integration of data and voice in one switching system. The computer simulation model is developed to analyze the data portion of the proposed system only.

The simulation is implemented using Simulation Language for Alternative Modeling (SLAM II). The network orientation of SLAM II is used to represent the system with a user written FORTRAN function provided to determine the transmission link rates.

A valid model is developed from a widely accepted analytical analysis of a star network similar to the proposed system at Heidelberg. The final validated model is

then exercised with two different packet switch rates and two different link speeds of the telephone switch-to-packet switch interface.

The simulation results show that with the originally proposed configuration, the packet switching rate was not a significant factor in throughput and delay measures. The proposed 9600 bits/second interface between the packet switch and existing telephone switch is the limiting factor of the two performance measures.

Among the recommendations is the suggestion that an interface with a higher bit rate be considered since this was the major limitation of the proposed system.

I. INTRODUCTION

Overview

As computer technology advances, more information is available at all organizational levels in business, manufacturing and the military. Low cost microcomputers and minicomputer systems have paved the way for local information processing within departments or communities belonging to larger organizations. Data communications networks provide resource sharing capabilities within these sub-organizations creating "information islands". The natural progression of communication technology is to integrate these islands of information into one organizational information system. This provides a more efficient flow of information between all the parts, creating a more cohesive organization.

Background

A requirement exists for the transmission of data and voice between 16 existing approved communications networks (COMNETs) at various levels of security within the Heidelberg information community at Heidelberg, Germany. Tasking for this requirement was assigned to the Assistant Director of Information Management, 1st Information Support Group, by letter dated 23 July 1986, from HQ, U. S. Army Europe and the Seventh Army. The U. S Army Information Systems Engineering Command - Europe is managing the project and is sponsoring this thesis.

The goal of implementing the COMNET interconnection backbone, as outlined in the Capability Requirements (CAPR) Document (9), is to provide a complete device-to-device data processing and communications network. Device-to-device is defined as personal computer-to-personal computer (PC-to-PC), PC-to-minicomputer, PC-to-mainframe, minicomputer-to-minicomputer, minicomputer-to-mainframe and mainframe-to-mainframe.

The basic user requirements outlined in the CAPR are:

a) To provide users with the ability to connect or disconnect from the backbone without any apparent interruption to the user of any other device operating on the system.

b) The backbone must provide users with the ability to interact with each other on existing COMNETs.

c) The backbone must provide for expansion as required.

d) The backbone must provide gateway capabilities to interface with other Local Area Networks (LANs) as required.

The sponsor plans to implement the Integrated Services Digital Network (ISDN) as the baseline architecture for the integration of data and voice in the backbone system. This is consistent with the present long-term United States Army ISDN policy for the integration of data and voice as stated in:

1. Department of the Army Policy Letter, 9 November 1984, subject: Policy on Army Base Communications.

2. HQ, USAISC Policy Letter 85-1, 16 September 1985, subject: Integrated Services Digital Network (ISDN) Policy.

3. HQ, USAISC Letter, 27 February 1987, subject: USAISC Office Automation Policies.

Based on the ISDN criteria, the goal is to achieve a data rate of 144Kbps at each desk terminal and move toward fiber optics as a network transmission medium. The system must support about 2620 devices of approximately eighteen different vendors. The backbone will encompass a distance of about 1.5Km within the confines of the Campbell Barracks and 7.5Km community wide.

Problem Statement

Headquarters, U. S. Army Information Systems Engineering Command - Europe expressed a desire for a computer simulation model to be constructed and exercised to determine the feasibility, throughput and delay characteristics of the proposed COMNET backbone.

Purpose

The purpose of this thesis project is to design and analyze by simulation, a model of the proposed communications backbone that will provide interconnectivity of data and voice for the 16 existing independent COMNETs at the Campbell Barracks in Heidelberg, Germany. The backbone must

provide gateways to connect with other local area networks as required. The Integrated Services Digital Network will be used as a basis for the design. The sponsor's proposed topology is a fiber optic star to allow integration with the existing telephone system, that provides concentrators and distributed switches for future growth.

Summary of Current Knowledge

Integrated Services Digital Network. Conrad described ISDN as a modern digital system aimed directly at replacing the present analog telephone system which carries virtually all data communication traffic (4:62). ISDN provides its users with various services and calling features across integrated subscriber loops from central offices by way of a high speed digital path for data, facsimile, videotext and signaling over the same connection as voice. These services can be used either separately or simultaneously by multiplexing over the same lines(3:42).

From a users perspective, ISDN offers services, protocols and performance. From a control perspective, it provides flexible and dynamic connection and call processing that makes the many integrated services available to the user. Therefore, a model of ISDN should allow the representation of the various services seen from the user's perspective (11:157).

In developing ISDN protocol and architecture models, Potter stated that the generic pieces to any model of teletraffic studies should include the following:

a) underlying services requested by users which produce the input traffic demand distribution,

b) the signaling control provided by the network or user for

i) connections - needed for the transport of the demand traffic, the routing, availability of resources as well as guaranteeing a request or expected grade of service,

ii) calls - since ISDN provides user controlled functionality, flexibility of possible configurations and versatility of access to the network or remote data bases currently being designed into both ISDN access protocols and internetwork protocols,

c) the implicit level of performance of the signaling network, protocols, network switches, relay points, data bases, internetworking units and resource availability,

d) the cost to provide a particular level of service. (11:157)

Potter noted that before an appropriate ISDN model can be developed, various information flows (user and signaling information) must be determined (11:161). In ISDN the information flows over three channel types: B, D and H. Over these channels, ISDN provides packet and circuit switching capabilities (15:144-151).

Conrad (4:62) examined the set of ISDN interfaces to provide communication between local as well as non-local users (see Figure 1). They are the R, S and T interfaces. The R interface provides traditional RS-232 or equivalent non-ISDN connections. The S and T interfaces are ISDN digital interfaces with multiple B and D channels. He noted that European carriers provide the network termination equipment (NT1) and the network interface equipment (NT2). In the United States, the users are free to acquire their own network interface equipment. Therefore, the basic regulated service end point is at the T interface rather than the S interface as in Europe.

Conrad also noted that subscriber terminal equipment (ST1) is fully compatible with ISDN requirements and connects directly to NT2 equipment. ST2 equipment with interfaces like RS-232, V.24/V.28, X.21, etc. connect through the terminal adapter (TA) which provides protocol conversion. NT2 equipment provides protocol handling, switching, concentrating and multiplexing. This equipment typically interfaces with other networks and provides expansion capabilities for the system. NT2 relates to the link and physical layers of the Open Systems Interconnect (OSI) Model. NT1 provides the electrical termination to the customer. It is basically equivalent to the physical layer of the OSI Model (4:62).

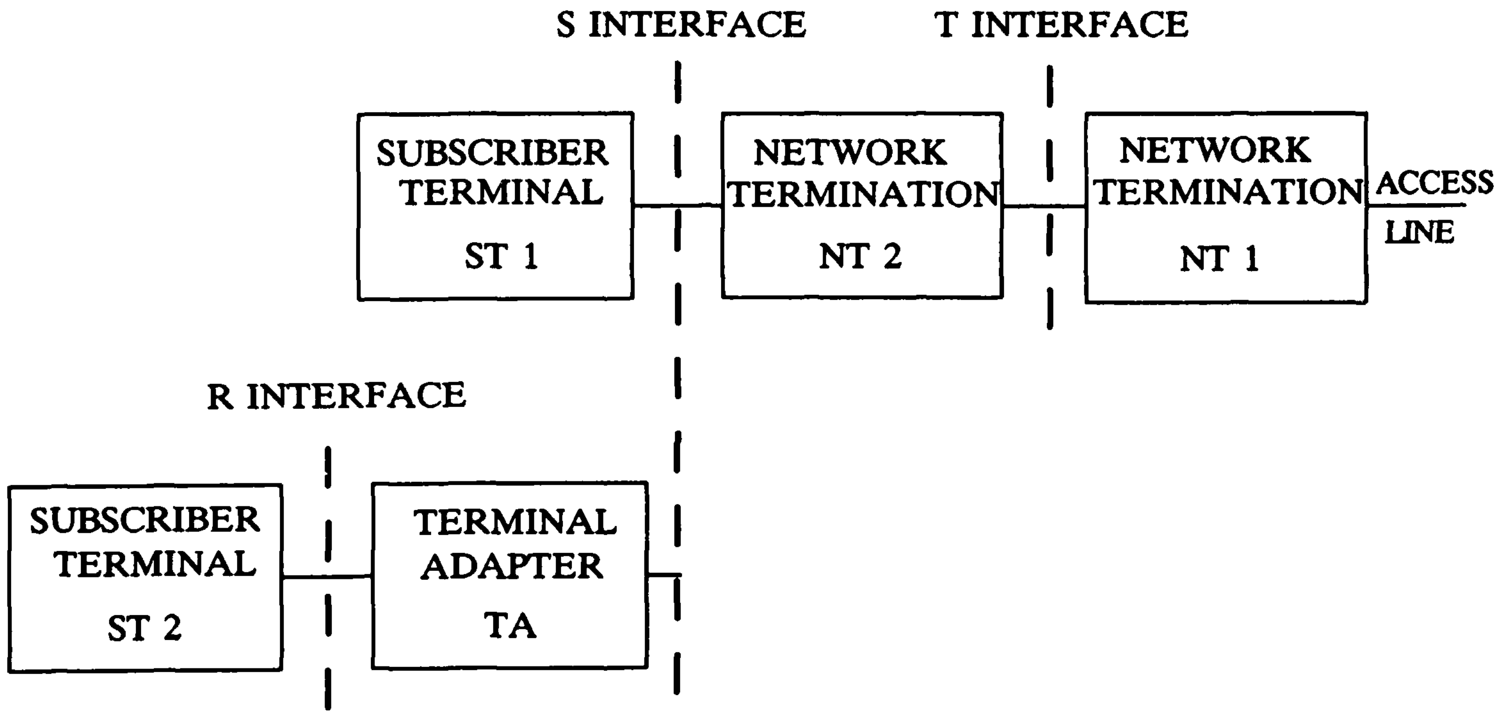


Figure 1. ISDN Functional Interface Group

In summary, ISDN is a central office communication system providing end-to-end digital connectivity with user control of service features. Integration of voice and data is provided as well as functional integration of packet and circuit switching. Connection to existing analog systems and digital communications networks is provided through common physical interfaces.

Simulation. A simulation is the imitation of the operation of a real-world process or system over time. Whether done by hand or on a computer, simulation involves the generation of an artificial history of a system, and the observation of that artificial history, to draw inferences concerning the operating characteristics of the real system (2:2). A simulation model in the form of a set of assumptions about the operation of a system is necessary to understand the behavior of the system over time. A model is composed of a collection of entities with associated attributes. Activities represent periods of time in a system. The state of a system is a collection of variables at any given time. Events in a system are the instantaneous changes of state (8:127).

Systems can be categorized as discrete or continuous, but few systems in practice are wholly discrete or continuous. However, since one type of change predominates for most systems, it will usually be possible to classify a system as being discrete or continuous (2:7). Variables

change at discrete points of time in a discrete system, where in a continuous system the variables are continually changing. A communications system is characterized as a discrete system due to the bursty, non-continuous nature of its traffic.

Banks and Carson pointed out that the simulation model is a substitute and a simplification of a system for purposes of study. These two authors noted that it is not necessary to consider all the details of the system; however, it must be sufficiently detailed to permit valid conclusions to be made about the real system. (2:9)

Simulations can be conducted manually or with the aid of a computer. Much insight can be gained from simulating small systems manually but real-world systems are normally rather large. Since the amount of data stored and manipulated is so vast, the runs are normally conducted with the aid of a computer. (2:11)

Banks and Carson also noted that computer simulation allows the state of a system to be changed in various ways. Discrete event driven simulations are based on a simulation clock. The clock advances until the next scheduled event. The user is free to vary the system variables, the number of entities, the relationship between entities, and the values of attributes of entities. Statistics collection in a flexible simulation program gives the modeler valuable information to iteratively test and optimize a proposed or existing system. (2:53)

Schoemaker pointed out that dangers of simulation for the modeler include

a) wrong designs which can lead to an unintentional wrong projection of the object system onto the target system to be simulated.

b) wrong coding leading to incorrect results.

c) wrong tools. A general purpose programming language can mean laborious progress while a special purpose language that has its own "view" of the world can imply intolerable constraints on the model.

d) no user/management involvement in the development and implementation process. (14:80)

That danger can be lessened or eliminated with the application of modern software engineering principles. Verification and validation principles must be included in the evaluation of the performance of the simulation model. Verification determines if the model is executing as was intended; in other words, is the modeler building the system right?

Validation determines if the model is a proper representation of the system; in other words, is the modeler building the right model? Pritsker states that validation of simulation models may be difficult, but not unreasonable because of the correspondence between model elements and system elements (12:12).

Scope

Security issues are not addressed in this thesis. The system is treated as unclassified in all modeling and simulation.

The simulation model represents the backbone that interconnects approximately 2620 devices distributed among the 16 COMNETs in the various buildings on location. Throughput and delay characteristics are determined from the simulation runs.

Approach

A simulation model, rather than analytical, was requested by the sponsor to provide a base model for the network as it grows and changes over time. It will also provide a convenient means to experiment with configuration changes as the network evolves in complexity. This thesis develops a valid simulation model of the backbone network that represents the actual system that is proposed at the Campbell Barracks. A computer simulation of the system is exercised to determine the throughput and delay characteristics. The computer simulation allows topology modifications of transmission links and packet switching rates to determine the optimal configuration.

The steps to accomplish these goals were:

1) Visit the Campbell Barracks to conduct a thorough requirements analysis based on accepted software engineering principles with the users of the proposed system.

2) Study network modeling and simulation techniques in general.

3) Design the network system model for the proposed network.

4) Conduct simulation experiments.

Simulation Techniques Study. A study of simulation was conducted to obtain a working knowledge of general system simulation. The study concentrated on discrete event simulation because of the non-continuous nature of computer network traffic. Additionally, a computer simulation language was selected. Features considered in the selection was ease of use, ability to accommodate user written modules, ability to model protocols, clarity of the generated statistics, and understandability of the simulated model when reading the code.

Network Modeling Study. A study of analytical and computer modeling was conducted to develop techniques that were used in the design a valid model of the actual backbone network. Proven analytical methods that apply to the topology of the actual network were researched. This research also included study of similar models that have already been validated, with the idea of modifying them to fit the constraints of this thesis.

Design of the Network. A valid detailed design of a computer network model was developed. This was done by first designing a model of a subset of the actual network.

Then performance characteristics, message delay and throughput for example, were manually calculated for the subnetwork with the analytical methods developed from the network modeling study. A subset was used because of the intractable task of manually analyzing a network of 2620 devices. By working with a small tractable problem, a computer simulation model could then be designed and its performance results compared with results of generally accepted mathematical methods. By designing the computer model to replicate the results of the analytical model, a high degree of confidence is achieved that the computer model is accurate. The simulation method used to develop the small subnetwork model was applied to the complete network problem to develop an overall valid solution.

Simulation Experiments. The experiments were conducted to determine throughput and delay characteristics of the complete network under various topology modifications and load variations. They were designed to exercise the network under controlled conditions according to requirements that the sponsor provided. One of the experiments was to study the effects on throughput of varying the packet switch speed with all other parameters held constant. Another was to study the effects of various link speeds in the network on system delay.

Thesis Organization

Chapter II presents the development of the model and the simulation methods that were actually used. A simulation language is determined based on its ability to provide a flexible and accurate representation of the system. In order to establish a high degree of confidence in the simulation model, analytical studies are conducted to determine its validity. Coding methodologies are presented for the program modules of the simulated system.

Chapter III presents the Heidelberg model and the detailed description of the code used to represent that model. Documentation of the code is also presented in the form of SLAM II network diagrams.

Chapter IV presents the experiments that were conducted to derive the characteristic delay curves for the backbone system.

Chapter V includes the thesis summary with conclusions and recommendations.

II. MODEL DEVELOPMENT

Introduction

The objective of this chapter is to develop the simulation model of the proposed Heidelberg system. The final simulation model is developed by first establishing a baseline model validated against known analytical results of a system similar to the one at Campbell Barracks. Then the validated baseline model is expanded to represent the actual proposed network.

The Heidelberg System

In order to validate the software model, it must be shown that the model accurately represents the actual system being modeled. The actual system at Heidelberg will be built around an ISDN oriented switch in a star configuration. Data traffic from terminals, hosts and other data devices will be transmitted to and be routed by the switch. Information collected during the requirements analysis site survey indicates that a phased approach will be used in building the final system.

In the first phase, a stand-alone packet switch will be installed near the existing European Telephone System (ETS) telephone switch gear. Only hosts in the Heidelberg information community will be connected in this packet switched network. This basically includes minicomputers (e.g VAX 11/XXX series or a machine of equivalent performance) and

mainframes. Some of those hosts have back end Local Area Networks (LANs) connected. Existing spare twisted pair will be used to interconnect approximately 20 hosts.

Phase 2 is planned to integrate the existing non-host oriented LANs into an ISDN oriented system. This will consist of an upgrade of the ETS by changing the analog subscriber loops to digital subscriber loops. The existing telephone "Key Systems" will be abandoned in favor of a single line concept for each digital telephone and data device. In a pure ISDN system the telephone and data terminal can be an integrated ISDN device. Otherwise, a non-ISDN data device or LAN gateway is interfaced with a terminal adapter (TA) and multiplexed to the ISDN switch via an NT2 device (also generically referred to as a gateway). This will require multiplexing over fiber optic links since the number of existing twisted pair is not adequate for a single line concept. Fiber optics will be used to be consistent with the U.S. Army policies stated in Chapter 1.

The packet switch will be interfaced with the ETS through a data link level connection. This interface will run the LAP-D link access protocol (1:1-12). This will allow access to the 20 hosts of the Phase I network from terminals in existing non-host LANs or ISDN data devices connected to the ETS. Only data channels, no voice, will be modeled in this thesis.

The Model

In addition to the 20 hosts that will be connected in Phase I, approximately 16 existing LANs of major organizations located at Campbell Barracks will be interconnected to the packet switched network via the ETS through gateways in Phase II. Therefore, a model of the data portion of a future ISDN oriented network will consist of 20 hosts connected directly to the packet switch and 16 gateway inputs from existing LANs (see Figure 2). As the network grows, the sponsor will be able to modify the model to reflect expected growth; for example, impacts of future ISDN data devices connected to the ETS when the number of these devices is known.

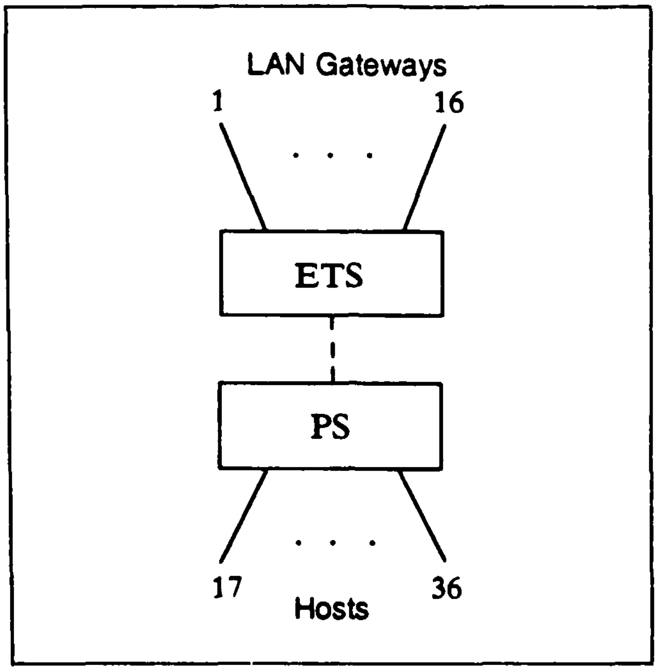


Figure 2. Phase II Network Configuration

Simulation Language

The computer simulation language chosen for for the model is SLAM II - Simulation Language for Alternative Modeling. SLAM II is a FORTRAN based language developed by Pritsker and Associates (12). It supports three views of modeling:

a) discrete event

b) continuous

c) network

The network view is used in this thesis. This view uses network symbols for building graphical models that are easily translated to code input statements for direct computer processing.

SLAM II was chosen because the graphical network symbols provide easy to understand documentation of program flow and simple translation to code. Also, future modification or expansion of the model will be simple due to the clarity of the logical network symbols as well as the ability of SLAM II to integrate user written FORTRAN event modules and user functions. This allows flexibility to meet future simulation needs of the sponsor or configuration changes to the network.

Classical Analytical Model

Much widely accepted analytical work in queueing theory and network analysis has been done by Kleinrock (10). He

represents a computer network as an unsteady flow of messages through a network of channels between nodes. This unsteady, or stochastic, flow means that the time between arrivals to the system and the demand placed on the channels by these arrivals are random quantities. Unlike steady flow, a queue may form during stochastic flow even if the average flow rate does not exceed the channel capacity. However, the queue length remains finite (10:5).

One of Kleinrock's investigations centered around the stochastic flow of messages through a network of communications centers. He noted that queues of messages form at the network nodes due to the sporadic message flow, thus requiring some form of storage at each node. His single most significant measure of performance is average message delay. This is the time for a message to travel from origin to destination. A traffic matrix whose entries, γ_{ij}, are used to describe the average number of messages generated per second, having an origin of i and a destination of j. The external traffic entering the network is given as $\gamma = \sum_{ij} \gamma_{ij}$ (see Figure 3).

The analytical model that he developed is based on several assumptions to provide mathematical ease and tractability. It is assumed that all node-to-node channels are noiseless and intranode delays are negligible. Traffic is considered to be data only with no telephone or direct wire traffic. Each message has a single destination and must

reach its destination before leaving the network. Each node is assumed to have unlimited storage capacity. Origin times and lengths of messages are random variables. Interarrival times between messages are exponentially distributed (i.e. Poisson process). Message lengths are also exponentially distributed.

	1	2	3	.	.	j
1	γ_{11}	γ_{12}	γ_{13}	.	.	γ_{1j}
2	γ_{21}	γ_{22}	γ_{23}	.	.	γ_{2j}
3	γ_{31}	γ_{32}	γ_{33}	.	.	γ_{3j}
.	.	.	.	.	.	.
.	.	.	.	.	.	.
i	γ_{i1}	γ_{i2}	γ_{i3}	.	.	γ_{ij}

Figure 3. Generalized Input Traffic Matrix

From queueing theory, the utilization factor is given as

$$\rho = \frac{\gamma}{\mu C} \qquad (1)$$

where γ is the total arrival rate of messages to the system from all external sources. $1/\mu$ is the average length of messages from all sources. C is the sum of all channel capacities in the network.

Each node in the network is represented as a single server queue (see Figure 4).

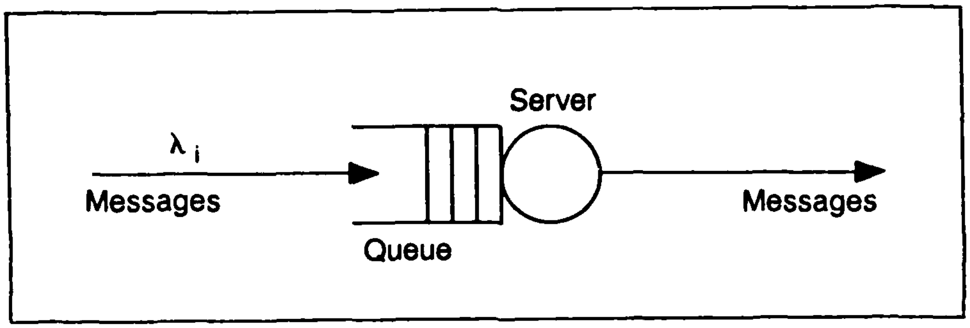


Figure 4. Single Server Queue

The Poisson interarrival rate at the ith node is λ_i messages per second. The average delay for the ith channel is

$$T_i = \frac{1}{\mu C_i - \lambda_i} \qquad (2)$$

The arrival rate of messages to all channels in the system is $\lambda = \sum_i \lambda_i$. The total averaged delay in the network is given by

$$T = \sum_i \frac{\lambda_i}{\lambda} T_i \qquad (3)$$

Kleinrock analyzed a five node network in a star topology having the channel capacities (bits/second) as shown in Figure 5 (10:19-33).

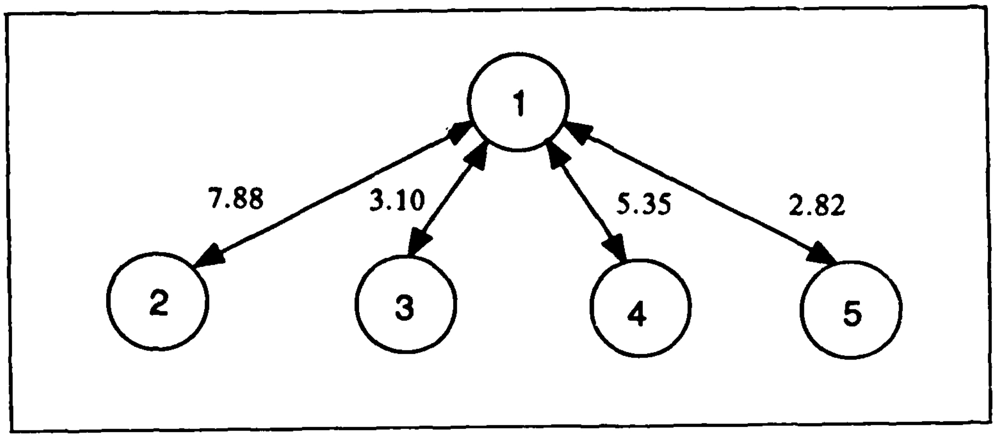


Figure 5. Kleinrock's Star Model

Table 1 shows the traffic matrix he used. This matrix represents a load of $\rho = 0.1$, $1/\mu = 0.1$, $C = 38.33$ and $\gamma = 38.33$. If ρ is allowed to vary, a delay curve is found using equations (2) and (3) (see Figure 6).

Table 1.

Kleinrock's Input Traffic Matrix

Node	Messages/Second 1	2	3	4	5
1	-	9.340	0.935	2.940	0.610
2	9.340	-	0.820	2.400	0.628
3	0.935	0.820	-	0.608	0.131
4	2.940	2.400	0.608	-	0.753
5	0.610	0.628	0.131	0.753	-

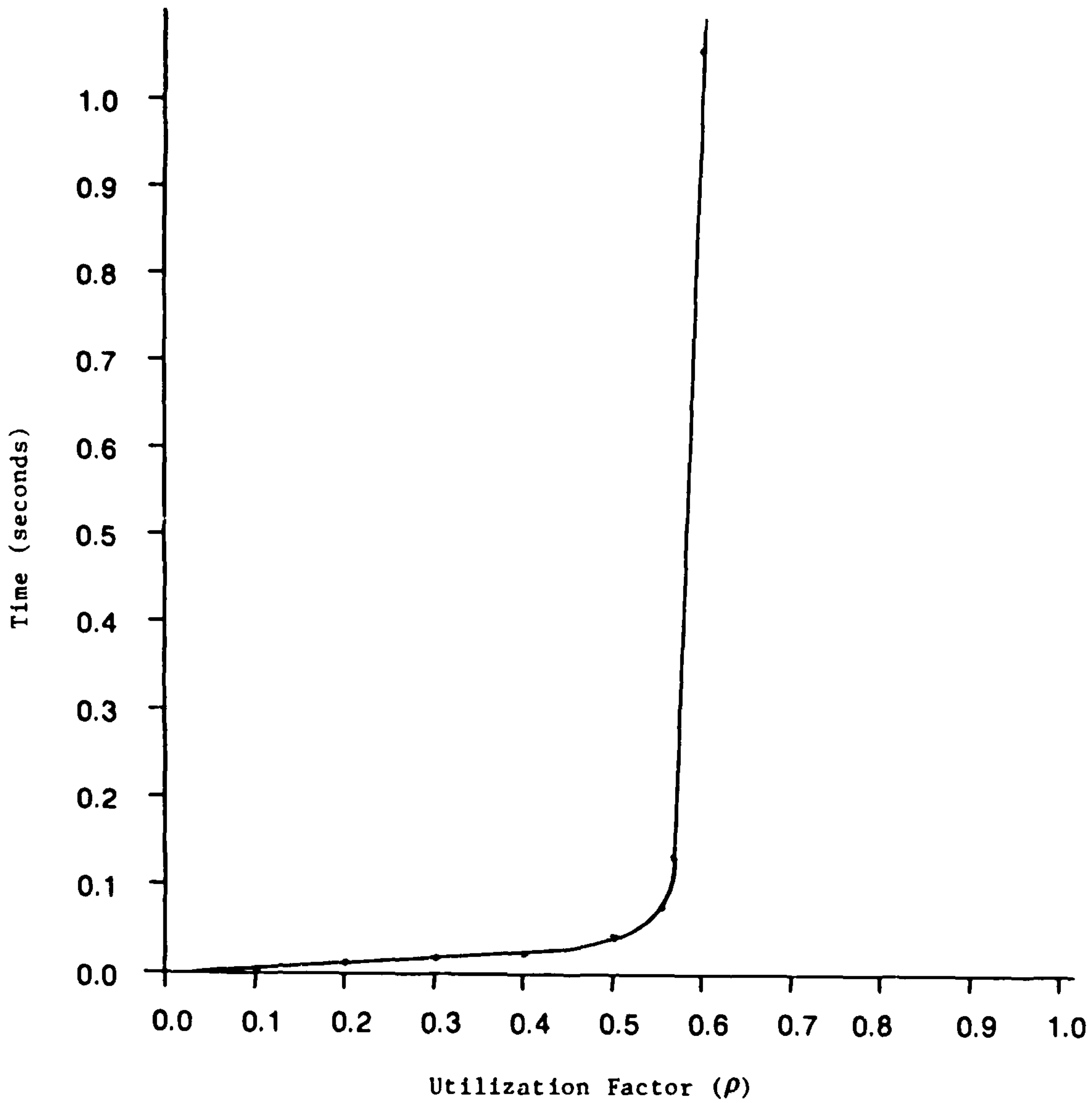


Figure 6. Average Delay Curve - Analytical Model

Baseline Model

A baseline SLAM II model was developed by Garcia (7) to simulate Kleinrock's five node analytical model. This model validates the programming technique and provides a basis to expand to the full simulation model with a high degree of confidence that it accurately represents the actual system.

The baseline model consists of four modules:

1. Message Generation
2. Node Processing
3. Node-To-Node Transmission
4. Statistics Collection

Figure 7 shows the SLAM II network diagram for the baseline model.

The message generation module uses the traffic matrix of Table 1 as input to the network. A create node is implemented for each γ_{ij} entry in the table. The time between creations is exponentially distributed with a mean of $1/\gamma_{ij}$ seconds. The assign node places the destination corresponding to γ_{ij} into attribute 2. An input queue represents each node inputting traffic to the system over one of four input channels to the central node.

The node processing module represents the central switching node only. Messages are routed to the proper node based on attribute 2. Messages destined for the central node are sent directly to the statistics module and terminated.

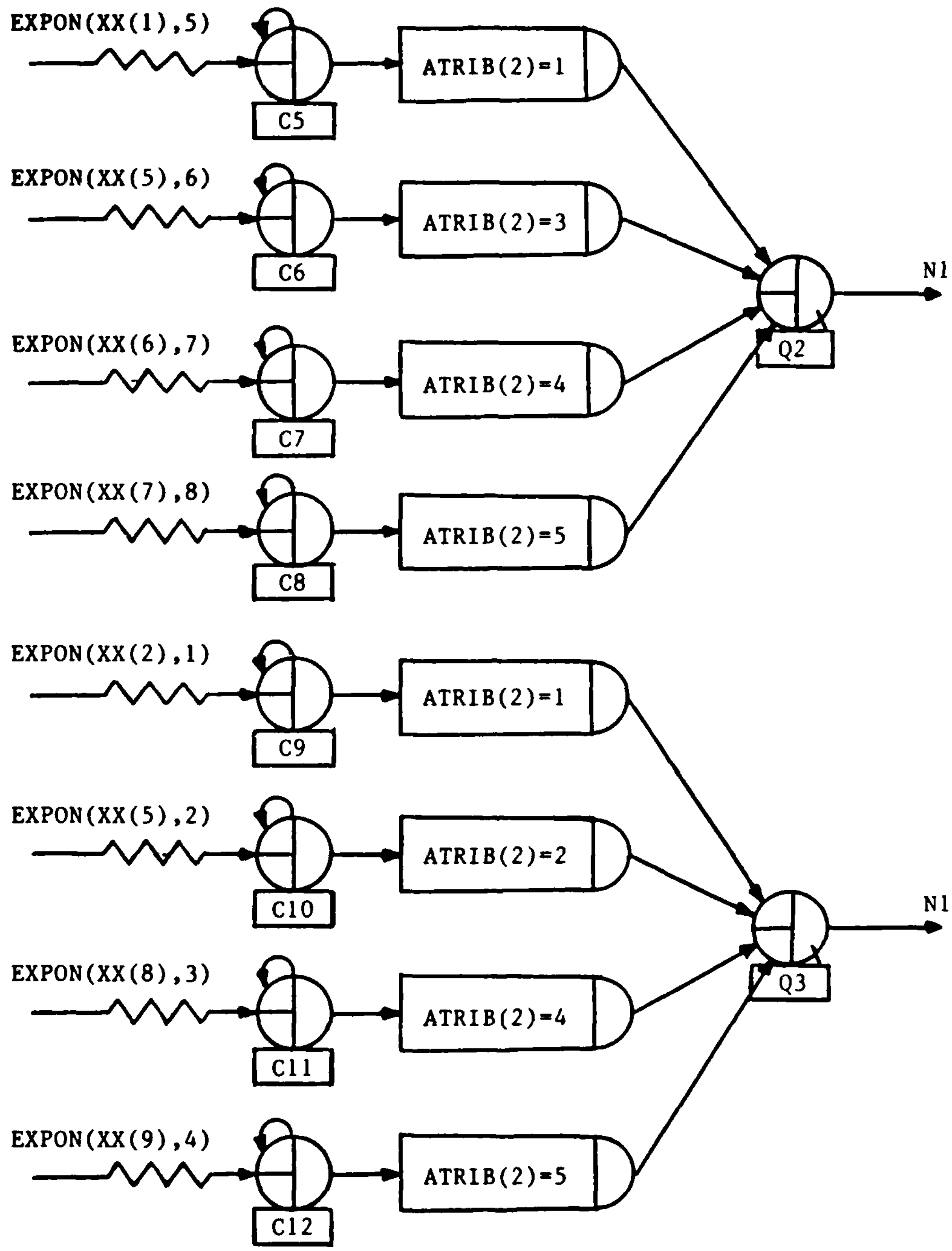


Figure 7a. Baseline Model Network Diagram (7)

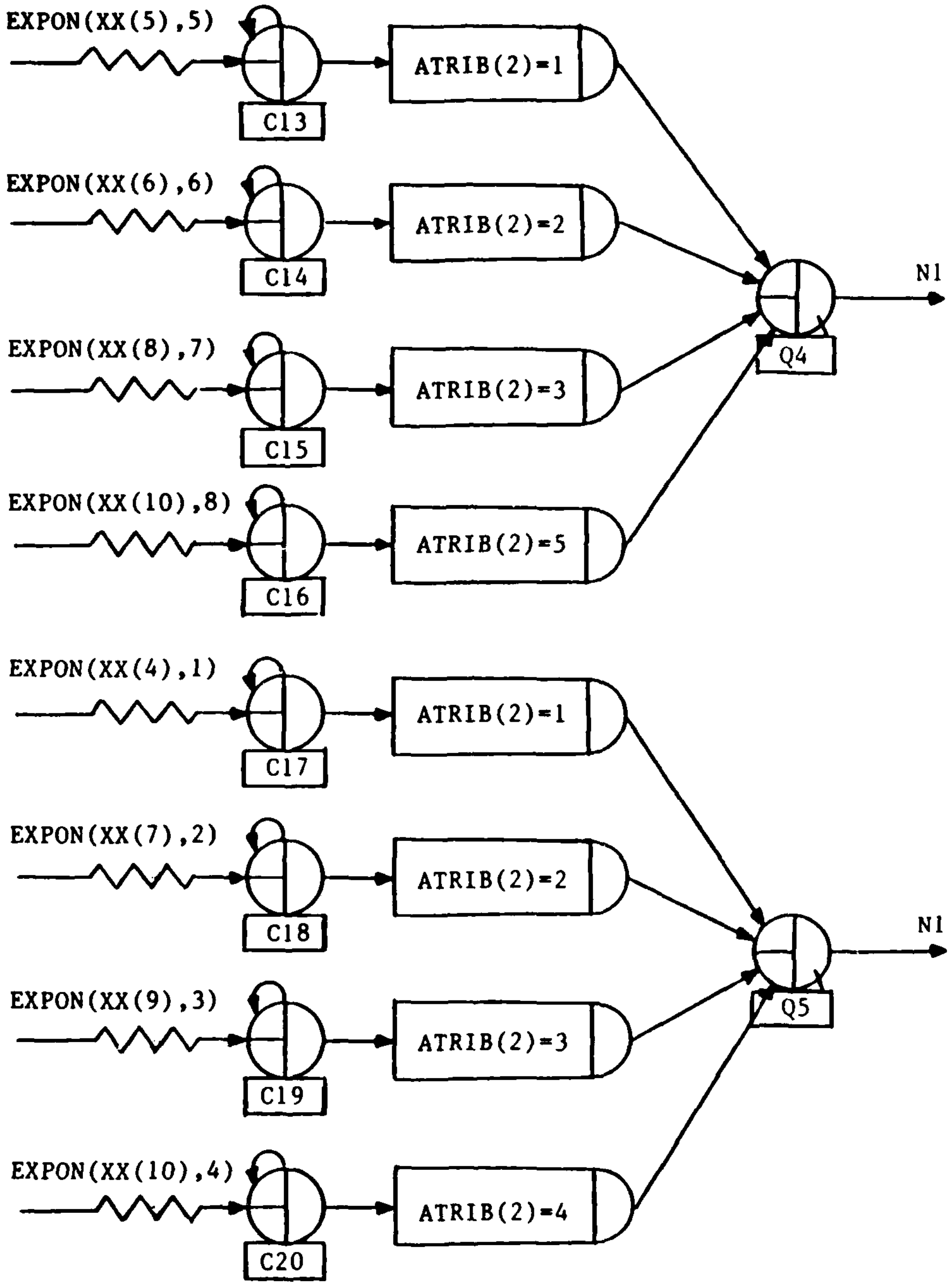


Figure 7b. Baseline Model Network Diagram (7)

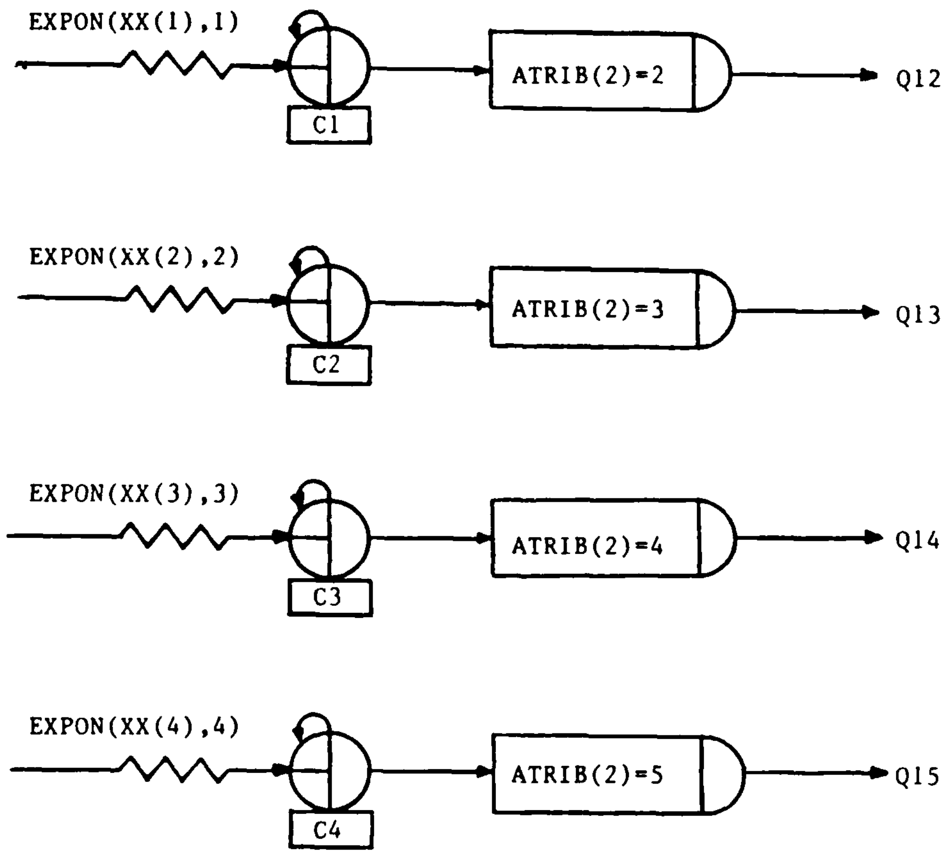


Figure 7c. Baseline Model Network Diagram (7)

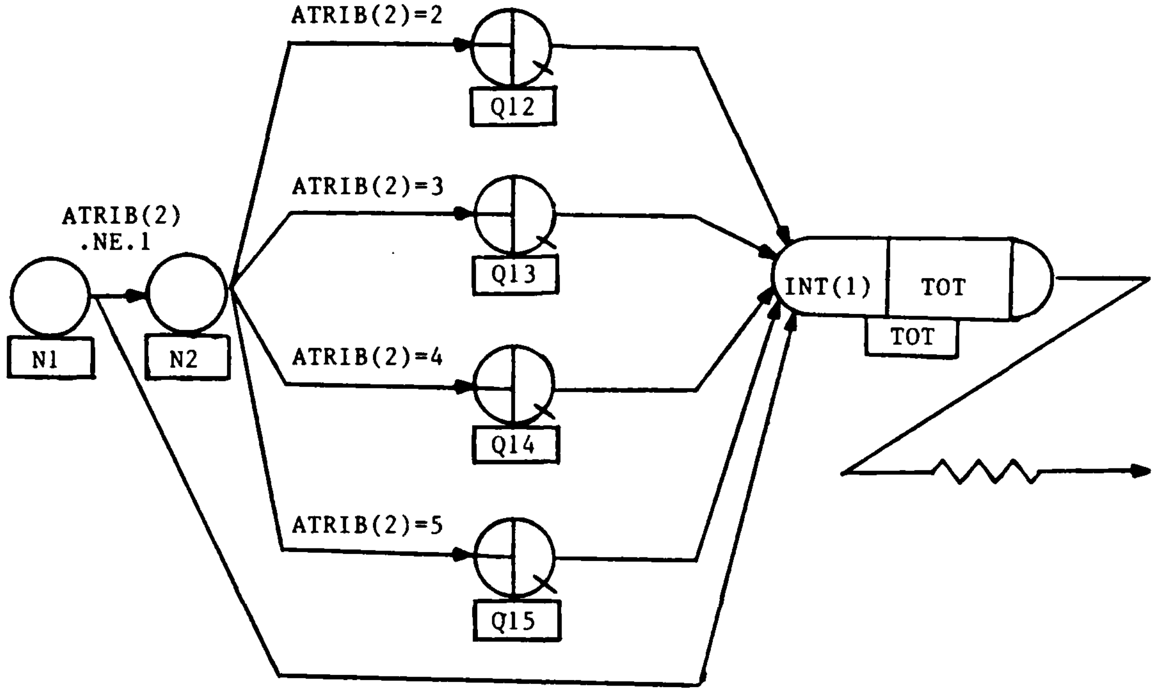


Figure 7d. Baseline Model Network Diagram (7)

The node-to-node transmission module consists of service activities of a duration of $(1/\mu)/C_i$ seconds. Each queue/service activity represents one of the four output channels from the central node.

The mean message delay is collected by the collect node TOT. Each entity's time in the system is recorded and the final total network mean delay is reported in the SLAM II summary report.

Figure 8 shows the system delay results of the SLAM II baseline model compared with those of Kleinrock's analytical model. Based on this comparison, it is shown that the baseline model is a valid basic design upon which to construct the full model.

Model I

Model I was developed from the baseline model to expand the channel links to include link transmission errors and a link level ACK/NAK protocol along the lines of the work done by Garcia. SLAM II queues were changed to await nodes and the channels represented as resources. This allows the ACK/NAK and error functions to complete while the next message in the queue waits for the channel resource to become available. A working five node Model I was completed that gave consistent results with the baseline model. The next step was to expand the model to the size and configuration of the proposed Heidelberg network.

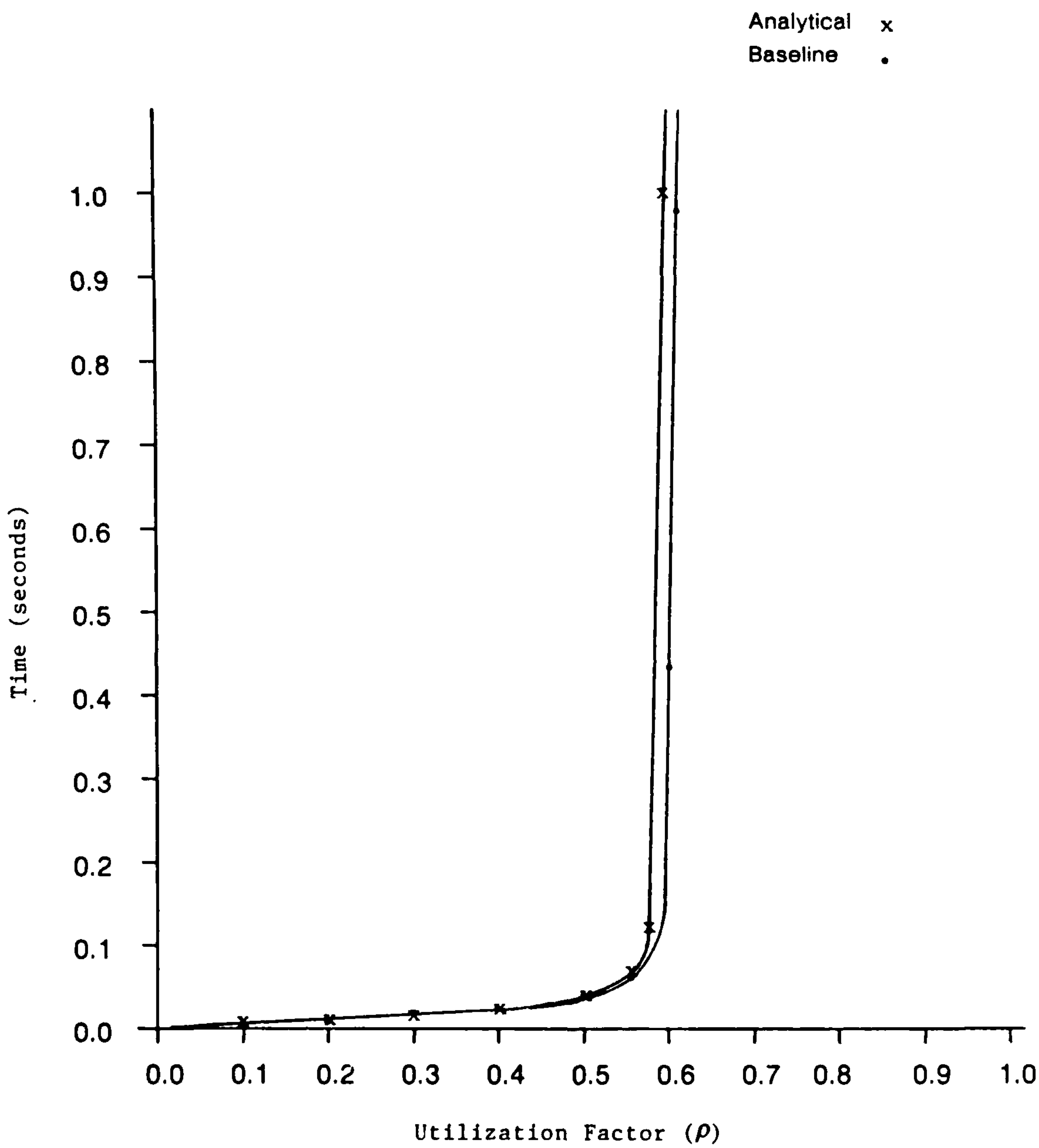


Figure 8. Baseline vs Analytical Delay Curves

The only basic difference between the Heidelberg system and Kleinrock's model is that the ISDN switching node contributes no original traffic and has intranode delay. Using Garcia's technique, one input queue is required for each node. Therefore, n nodes require n queues with n-1 create nodes per input queue (see Figure 7). The Heidelberg model has 36 input nodes. That would require 36 input queues with 35 create nodes per queue. That means 1260 create nodes must be coded to represent each non-diagonal entry of the input traffic matrix, minus those for the central node representing the ISDN switch since they contribute no external traffic.

Traffic Input. The expected traffic input to the Heidelberg network is unknown. Therefore, it was decided to assign equal input traffic rates, γ_{ij}, for each traffic matrix entry for an overall network traffic input of γ calculated at a given utilization rate of ρ. Each ρ corresponds to a value of packets/second input to the network. This will give a delay characteristic of the network at various loads. According to Kleinrock, a highly non-uniform traffic pattern will yield a minimum message delay (10:122). Since the expected traffic is unknown, a uniform traffic pattern should give worst case delay for the system under study, which is a desirable measure of performance. In light of this, each create node will have the same mean interarrival rate for its exponential distribution. Since

Poisson processes are additive, it was recognized that the message creation module could be redesigned with one create node representing the overall network traffic input. The mean packet inter-arrival rate is then $1/\gamma$. The source/destination pairs are now determined for each message from uniform distributions between 1 and 36.

Model II

The final working model, Model II, was designed from lessons learned in the development of Model I. The first version of Model II was built to verify the idea that one create node with a mean interarrival time of $1/\gamma$ is equivalent to n - 1 create nodes inputting n queues.

Again, the data of Table 1 was used to verify that Model II was equivalent to the baseline model. The redistribution of the data to make each γ_{ij} of equal value was obtained by dividing γ (the overall external input to the network) by the number of γ_{ij} entries; i.e. 38.33/20 = 1.916 messages/second at $\rho = 0.1$. The baseline model was used to develop a delay curve as ρ was varied.

Model II was set up with one create module representing inputs from all nodes except the central node. This represents 16/20 of the total network traffic input. Another create module representing input traffic from the central node contributes 4/20 of the total. Two modules were actually used since external traffic generated at the

central node is input to the network output queues while the other nodes input to the network input queues. The central node is again represented as an activity with no delay (see N1 in Figure 9).

Slam II allows multiple queues to be represented with a single await node. Each queue is associated with a link. The effect is equivalent to the discrete queues and activities of Figure 7. When the link is free, the proper path is selected by either source or destination. Figure 9 is logically equivalent to Figure 7.

The mean interarrival rate for input nodes is then $(20/16) \times (1/\gamma)$, and for the central node, $(20/4) \times (1/\gamma)$. The source values were determined from the uniform distribution of 2 to 5 for the input nodes. The source is always 1 for the central node. All destinations were determined from the uniform distribution of 1 to 5. Again ρ was varied and Model II was used to develop a new delay curve. Figure 10 shows that the two curves are nearly identical. Therefore, it is shown that Model II is accurate and that the method of using interarrival rates based on γ, or a proportion of γ if more than one module is used, is equivalent to one create node for each entry of the input traffic matrix. This will drastically reduce the amount of source code and keep the model compact. It still provides flexibility because if some individual channels must be modeled separately, they can be represented with separate

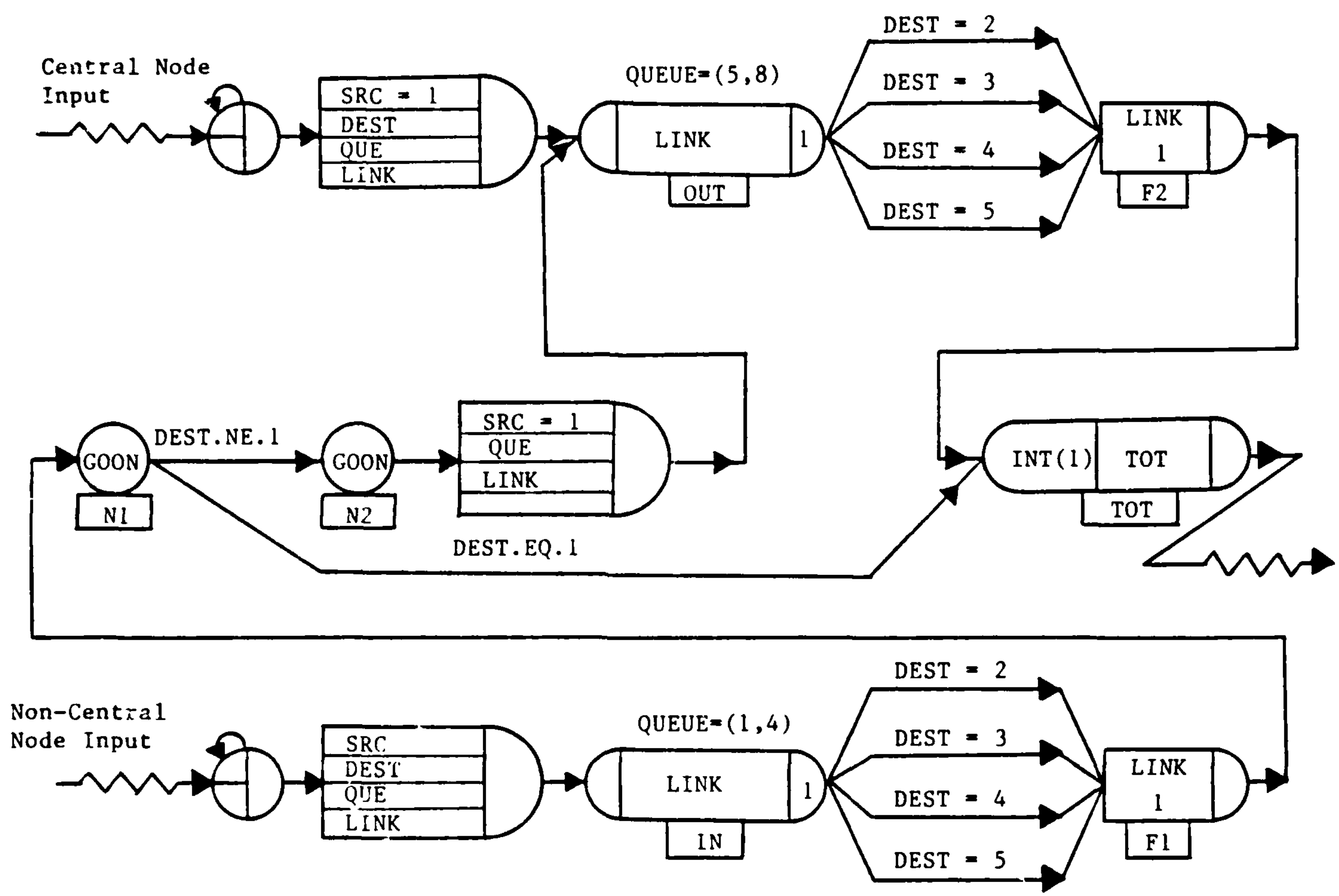


Figure 9. Model II SLAM II Network Diagram

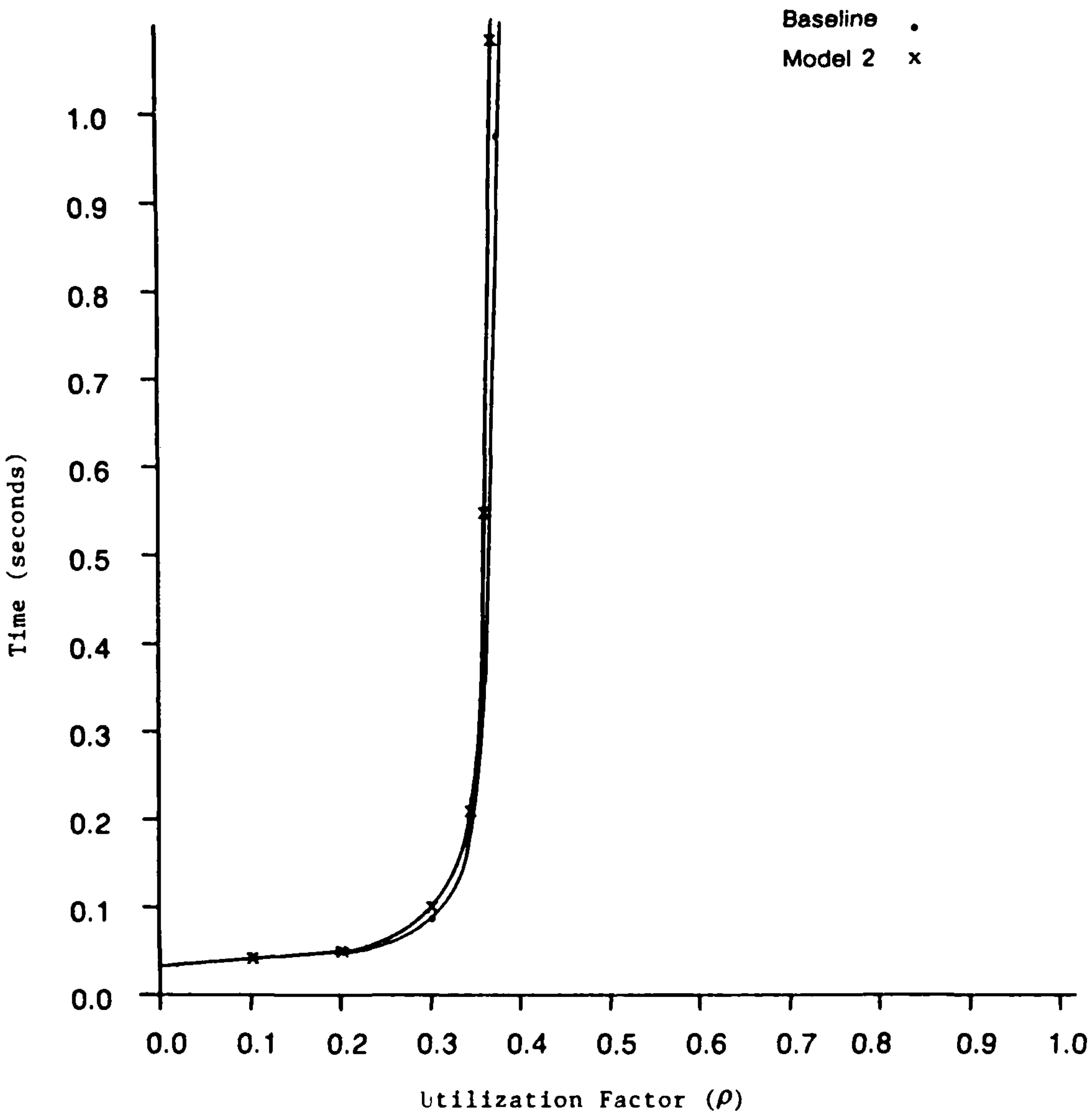


Figure 10. Model II vs Baseline Delay Curves

create modules with the interarrival rates adjusted proportionately across all inputs.

Summary

This chapter explained the process that was used to develop a simulation model that was based on widely accepted analytical work done on a network similar to the proposed communications backbone at Campbell Barracks in Heidelberg. A programming technique for a simulation model was found that accurately simulated the analytical model thus providing a basis for validating the model. That simulation model was modified to minimize the amount of source code and to more efficiently represent the type of data to be run in the simulation. This keeps the program manageable and simple while maintaining a high degree of confidence that it accurately represents the actual system. The results of the modification were verified against the baseline model for accuracy. The final product is Model II which was validated and verified against accepted analytical results.

III. THE HEIDELBERG MODEL

Introduction

Using the concepts developed in Chapter 2, Model II is expanded to represent the COMNET backbone at Heidelberg. This chapter describes variables and attributes used in the code, the data flow diagram that represents the model and a description of the SLAM II code that is derived from the network diagrams. The SLAM II network diagrams are contained in Appendix 2.

Variables and Attributes

Table 2 lists the SLAM II variables and with functions.

Table 2

SLAM II Variables

Variable	Function	
XX(1)	gateway interarrival time	(seconds)
XX(2)	host interarrival time	(seconds)
XX(3)	mean packet size	(bits)
XX(4)	minimum packet size	(bits)
XX(5)	maximum packet size	(bits)
XX(6)	ETS-PS link rate	(bits/sec)
XX(7)	ACK/NAK packet size	(bits)
XX(8)	error rate	
XX(9)	number of hosts	
XX(10)	packet switch rate	(bits/sec)
XX(11)	number of channels + 2	
XX(12)	ETS overhead	(seconds)

A detailed description of each variable is contained in Appendix 1.

Table 3 lists the attribute values for each entity that traverses the network.

Table 3

Attributes

Attribute	Function
1	creation time
2	destination
3	source
4	channel number
5	queue
6	packet length
7	transmission rate

A detailed description of each attribute is contained in Appendix 1.

Code Description

The data flow diagrams of the Heidelberg model are shown in Figure 11. They illustrate the packet flow in the four modules in which the actual SLAM II code is organized. The modules are:

a) Input Module

b) Transmission Module

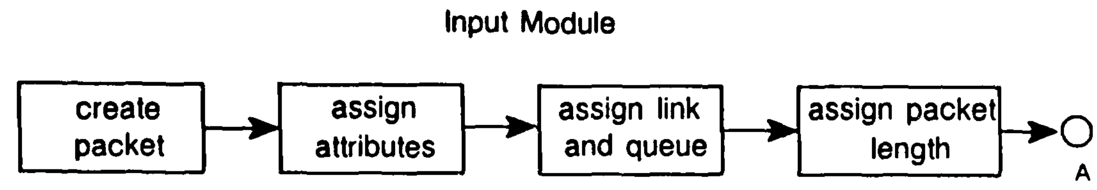
c) Switch Processing Module

d) Output Module

The actual SLAM II network diagrams of the modules from which the SLAM II code is derived are illustrated in Appendix 2.

Input Module. The input module begins with a create node inputting entities with exponentially distributed interarrival times. Gateway input to the ETS is represented with one input module. Host input to the packet switch is represented with another. This was done for ease of representing different channel transmission rates for hosts and gateways. A source/destination pair is then determined with two uniform distributions with different seed streams. Based on these source/destination values, an input queue, represented by an await node, and a channel, represented by a resource, are selected. The packet length is determined from an exponential distribution with XX(3) being the mean value. A range of packet sizes can be enforced by using XX(4), the minimum value, and XX(5), the maximum value. The packet is then assigned a transmission rate by USERF(1) and stored in RATE. It then awaits in its appropriate queue for transmission over a free channel.

Transmission Module. The transmission link from a gateway to the central node is represented as shown in Figure 11. When the transmission link is free the errorless link sends an entity directly to the FREE node at the transmission speed of LENGTH/RATE. However, when transmission errors are modeled, the error branch is taken a percentage of the time corresponding to the transmission error rate. The NAK is sent and another probability branch is taken. When a good packet is eventually sent, the branch



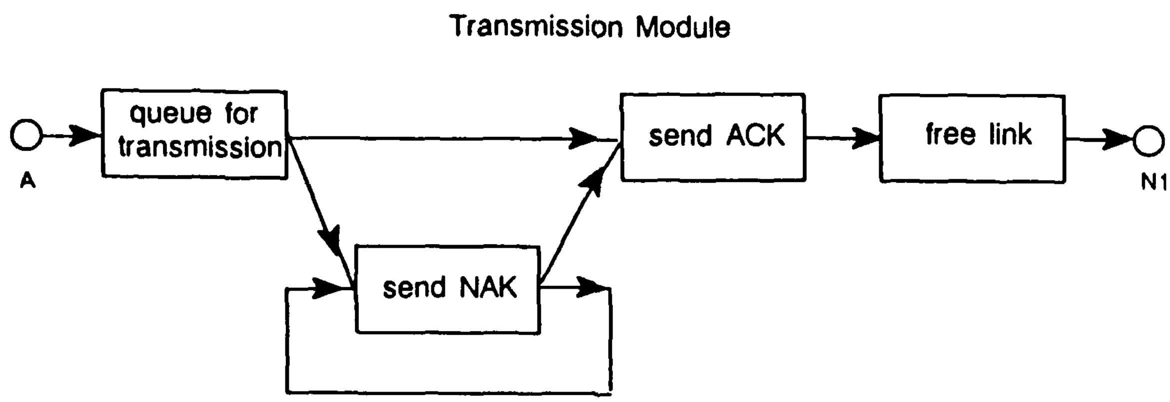


Figure 11. Data Flow Diagram of Full Model

Switch Processing Module

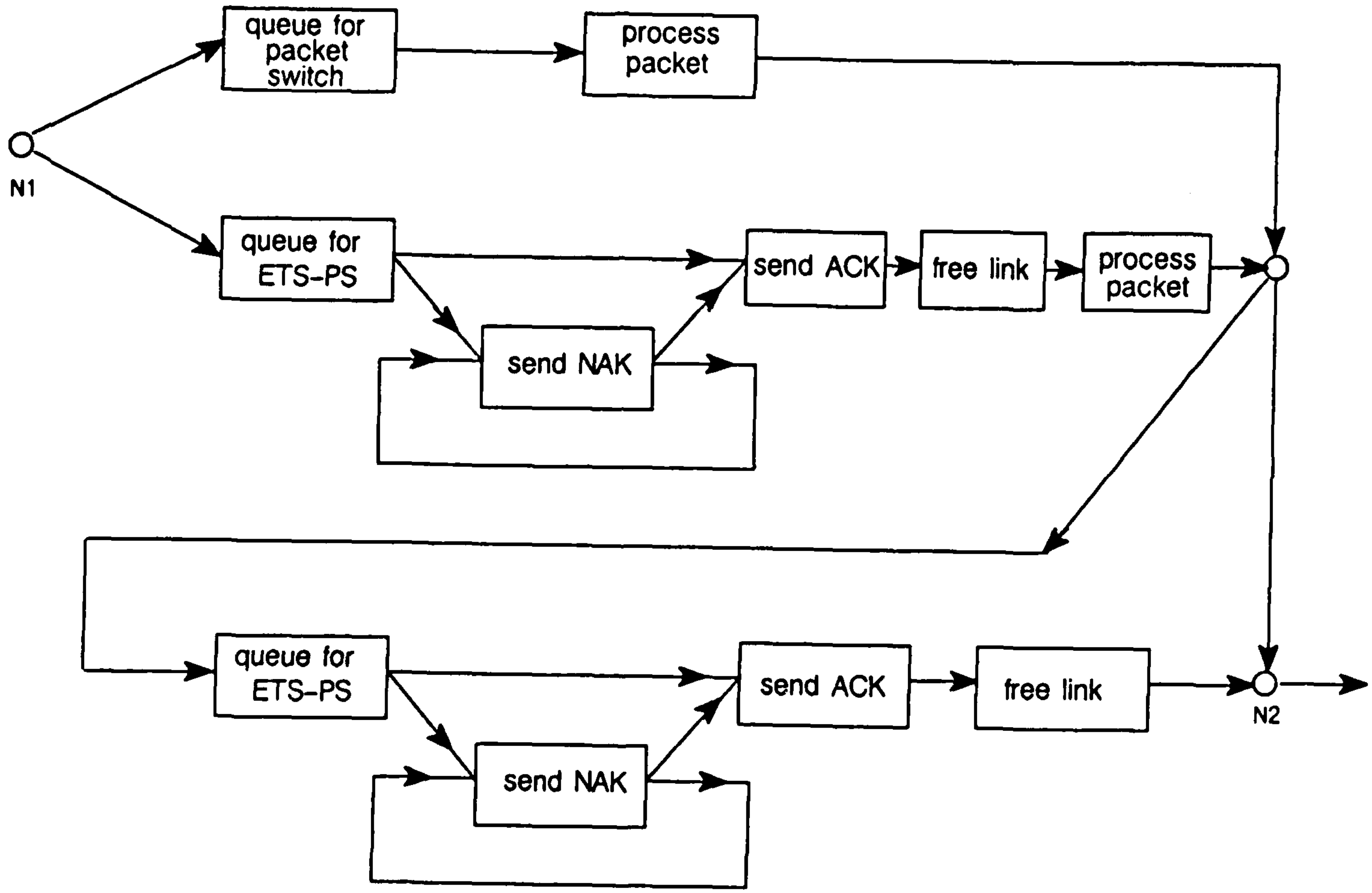


Figure 11 (cont). Data Flow Diagram

Output Module

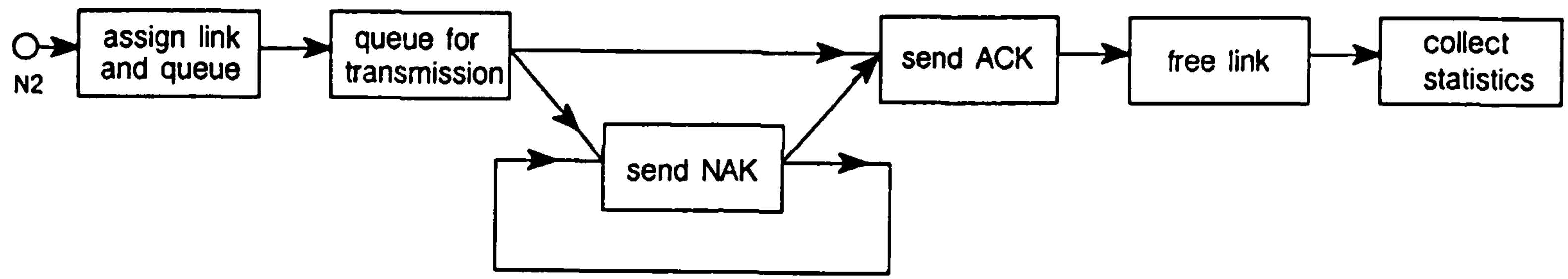


Figure 11 (cont). Data Flow Diagram

is to complete the transmission at the transmission speed of LENGTH/RATE. The link is then freed to allow the next packet transmission. The ACK/NAK packet size is held in XX(8). The transmission error probability branch decision is based on the contents of XX(9). The transmission rate is determined in USERF(1) and stored in RATE. This rate depends on whether a packet is transmitted from a gateway or a host. When the transmission is complete, the link is freed for transmission of the next packet in the queue.

Switch Processing Module. The switch processing module of Figure 11 represents the ETS, the link between the ETS and packet switch (ETS-PS link), and the packet switch overhead. The ETS processing overhead is assumed to be negligible compared to the combined processing overhead of the ETS-PS link and the packet switch. When the entity representing a packet enters the switch processing module at node N1 from a transmission module, a decision is made, based on the source, to determine if it is assigned to the packet switch queue directly or if it must traverse the ETS-PS link first. The ETS-PS link operates the same as the transmission link. After the packet switch processing, the destination attribute DEST determines if the ETS-PS link must be traversed in the case of the destination being a gateway. Otherwise, the entity is routed directly to the output module via N2 in the case of a host destination.

The ETS-PS link is interfaced to the two switches by the LAP-D protocol which is a version of the High Level Data Link Control (HDLC). This protocol allows many bidirectional logical links across one interface. It assigns each logical link an identity and provides each with transmission error recovery and flow control (1:1-12). This link was therefore represented as other channel links providing the low level error recovery.

Output Module. The source designation from the switch is always a 1 and is assigned to SRC. The destination queue and channel are derived from DEST. The channel transmission rate is determined from USERF(1) based on DEST. The entity representing a packet waits in the assigned output queue until the appropriate channel is free. Again, the transmission error probability branch decision is determined by XX(9) as in the transmission module.

The collect node STAT calculates the mean delay for the network by averaging the differences of an entity's creation time in attribute 1 and the current time when any entity arrives at the collect node. The entity is then terminated after the delay is calculated.

USERF(1). This FORTRAN user function is used to make logical decisions in determining the particular channel transmission rates based on the source or destination. This function can also be used to make other logical decisions that can't be done efficiently in SLAM II code. This could be of importance in future model expansion.

Entities. It was experimentally determined with a pilot run, that at least 100,000 entities must be collected to get a reliable delay curve. The horizontal portion of the curve can be consistently reproduced with various seed streams for the input distribution with less than 10,000 entities. However, as the queue lengths become longer with higher loads, the variability of the average waiting time is greater for each entity and so it necessary to collect the larger amount in order to reproduce the vertical portion of the curve that is consistent between runs. A collection of 150,000 entities did not significantly improve the consistency, so 100,000 was chosen.

IV. EXPERIMENTS

Introduction

The objective of this chapter is to explain the experiments that were run to exercise the model and to show their results. The results will characterize the network performance as a consequence of varying several network parameters that define the network. These parameters include channel transmission rate, ETS-PS link transmission rate and packet switch processing rate.

It was decided, in agreement with the sponsor, to concentrate this effort on the analysis of the the characteristics of the COMNET backbone. Other aspects of the ISDN system can be studied with subsequent efforts.

This chapter explains the experimental simulation runs in two parts. The first part contains results of two runs conducted on the Phase I model. The first run used a packet switch rate of 160 packets/second. The second was run with a packet switch rate of 400 packets/second. All other parameters were held constant.

The second part explains the three series of simulation runs that were conducted on the Phase II full model. The first series was modeled with the ETS-PS link transmission rate at 9.6K bits/second. Runs were then conducted with packet switch rates of 160 packets/second and then 400 packets/second. The second series was conducted in the same

manner as the first with the only difference being the ETS-PS link rate of 64k bits/second. A third series of runs was conducted to show the effects of various transmission error rates on system delay.

General Assumptions

At the time this thesis project began there were several unknowns about the proposed system. Therefore, reasonable assumptions were made to fill the gaps and proceed with the analysis from a realistic starting point based on input from the sponsor.

The packet switch computer was expected to be in the class of the Bolt Beranek and Newman (BBN) C/30. Determining throughput of a computer can be a vague and even controversial matter. Nevertheless, for purposes of this model the metric for throughput is packets/second. The C/30 processes approximately 160 packets/second with an average packet size of 750 bits in the ARPANET applications (6:38). This throughput value was used in the model.

The packet sizes are exponentially distributed, based on Kleinrock's assumptions, with a mean of 750 bits. The maximum and minimum limits of 1024 bits and 256 bits respectively are used as the packet size range (13:83).

In order to model the effects of the packet switching rate on the network, a BBN C/300 was also used in the model as a comparison against the C/30. The C/300 switches 400 packets/second with an average packet size of 750 bits.

The ETS-PS link transmission rate was given by the sponsor to be 9600 bits/second. At first glance, it appeared that this could be a potential bottleneck in the system. Therefore, the sponsor was contacted to verify this link rate. An affirmative reply was followed up with a copy of Defense Data Network, Installation, Test, and Acceptance Guide (5). In that reference, rates of several interfaces for the C/30 were researched. It was found that five basic inter-switch trunk rates are used in Defense Data Network applications using the C/30. They are 9.6K, 19.2K, 50K, 56K and 64K, all in bits/second (5:2-9). In light of this added information, it was decided to bracket this range by modeling the 9600, being the slowest, and the 64K, the fastest, to show the effects of the ETS-PS link transmission rate on the system.

The host transmission channel speed was given to be 64K bits/second. This rate was used in all experimental runs. The channel transmission rate on the ETS side of the switch was given to be 144K bits/second which corresponds to a 2B+D ISDN channel. The greatest number of devices in the COMNETs will probably be terminals typically operating at 9600 bits/second. A D channel would be capable of carrying some of that traffic as well. Therefore, experimental runs were included to show the network characteristics with D channel transmission rates at the same capacity as the 2B+D channel.

The expected network input traffic load was an unknown parameter. Therefore, the traffic input from each source was made equal within its own class of traffic, i.e. host or terminal. The total network capacity is the sum of all individual channel capacities. In the case of the full system, all host channels were assumed to be input with equal traffic intensity and all gateway channels input with equal traffic intensity. These two classes of input were proportioned according to contribution each class made to the total network traffic input. For example, if host channel capacity is 80% and gateway channel capacity is 20% of the total network capacity, then the input traffic intensity is proportioned accordingly.

Each simulation of a given configuration consists of three runs with different seeds for the input distribution. Each plotted point of the network delay graphs is the average of the three runs for a given point. In all runs, the difference between any one point and its average was found to be less than 1% in the active region of the curve to the left of the knee. This low error is expected in this region and is convincing evidence that the simulator is consistent. A warmup period of 100 seconds was used, after which the statistics were cleared.

The simulation runs on a VAX 11/785 for about 55 CPU minutes per data point when collecting 100,000 entities (see Chapter 3 for reason of this number of entities).

Host-to-Host Network

As described previously, Phase I will provide packet switched interconnectivity between hosts in the Heidelberg information community. Figure 1 illustrated that these hosts will connect directly to the packet switch. The first simulation run was made with a packet switch rate of 160 packets/second, a channel transmission rate of 64K bits/second and a mean packet length of 750 bits. Calculations for the packet interarrival rate ($1/\gamma$) at $\rho = 0.1$ are:

$$C = 64000 \times 40 = 2560000 \text{ bits/sec}$$

$$\mu = 0.0013333 \text{ packets/bit}$$

$$\gamma = 2560000 \times 0.0013333 \times 0.1 = 341.33 \text{ packets/sec}$$

$$1/\gamma = 0.002929694824 \text{ seconds/packet}$$

where C is the sum of all channel capacities, $1/\mu$ is the packet length and ρ is calculated from Equation (1) in Chapter 3.

The simulation results in Figure 12 show that the knee occurs near $\rho = 0.05$. The total network input traffic of 341.33 packets/second occurs at a utilization factor of 0.1. Therefore, a utilization factor of 0.0468 represents a traffic load of 160 packets/second. Note that this is the packet switching rate of the C/30. The average delay can be expected to be less than 0.08 seconds while the traffic intensity remains below the packet switch capacity.

A similar experiment was run with a C/300 as the packet switch. The results are consistent with those run with the

C/30. The knee occurs near a utilization factor of 0.1. The utilization factor of 0.117 represents a traffic load of 400 packets/second.

The delay curves of Figure 12 show that throughput is increased, but the delay is not significantly decreased with a faster packet switch. In both cases the delay will be less than .08 seconds and is not significantly different between curves in the active ranges below the knees.

Full System

The characteristics of the full configuration are illustrated in Figures 12 and 13. The simulation runs were designed to show the effects of the ETS-PS link transmission rates at the two packet switching rates.

The interarrival rates ($1/\gamma$) were calculated from the following values at $\rho = 0.1$, where t subscripts indicates terminal traffic on the ETS side and h subscripts indicate host traffic on the packet switch side:

$$C_t = 144000 \times 32 = 4608000 \text{ bits/sec}$$

$$C_h = 64000 \times 40 = 2560000 \text{ bits/sec}$$

$$\mu = 0.001333 \text{ packet/bit}$$

$$\gamma = 955.73 \text{ packets/sec}$$

$$\gamma_t = 614.4 \text{ packets/sec}$$

$$\gamma_h = 341.33 \text{ packets/sec}$$

$$1/\gamma_t = 0.0016276041 \text{ seconds}$$

$$1/\gamma_h = 0.0029297161 \text{ seconds}$$

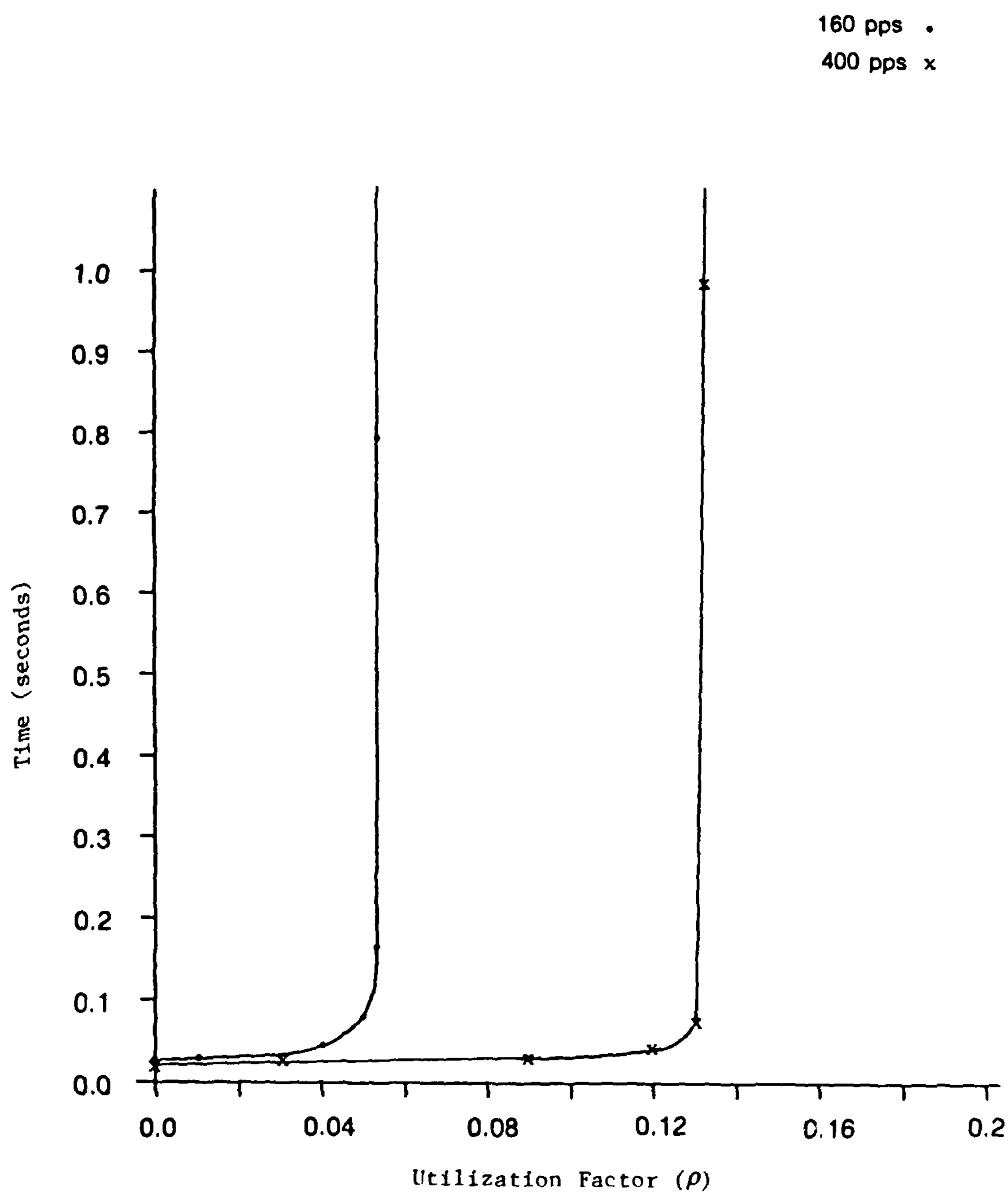


Figure 12. Phase I Average Network Delay

The terminal capacity is 64.29% and host capacity is 35.71% of the total network capacity. Therefore the total arrival rate, γ is proportioned as such to arrive at the given values of terminal and host interarrival rates.

Run 1. Figure 13 shows the configuration with a 9600 bits/second ETS-PS link and terminal channel transmission rates of 144k bits/second. This link is obviously the slowest component in the system. The knee occurs near a utilization factor of 0.001. The corresponding utilization factor for 9600 bits/sec is 0.00133 or 12.9 packets/second. Therefore it is shown that throughput is limited by the slowest component in the system, i.e. the ETS-PS link. As can be expected, a faster switch will not increase the throughput in this configuration and the curves verify this. The average delay can be expected to be below 0.18 seconds while operating below a utilization factor corresponding to the ETS-PS link rate. There is no significant delay improvement by implementing the C/300 switch.

Run 2. The capacity assignments, channel transmission rates and interarrival rate for Run 2 are identical to Run 1. The only difference is the ETS-PS transmission rate which is modeled at 64000 bits/second. Figure 14 shows the delay curve of this configuration. Based on Run 1, it would be expected that the knee of the delay curve would occur near a utilization factor in this configuration corresponding to 64000 bits/second, which is 0.0089 or 85.33 packets/sec.

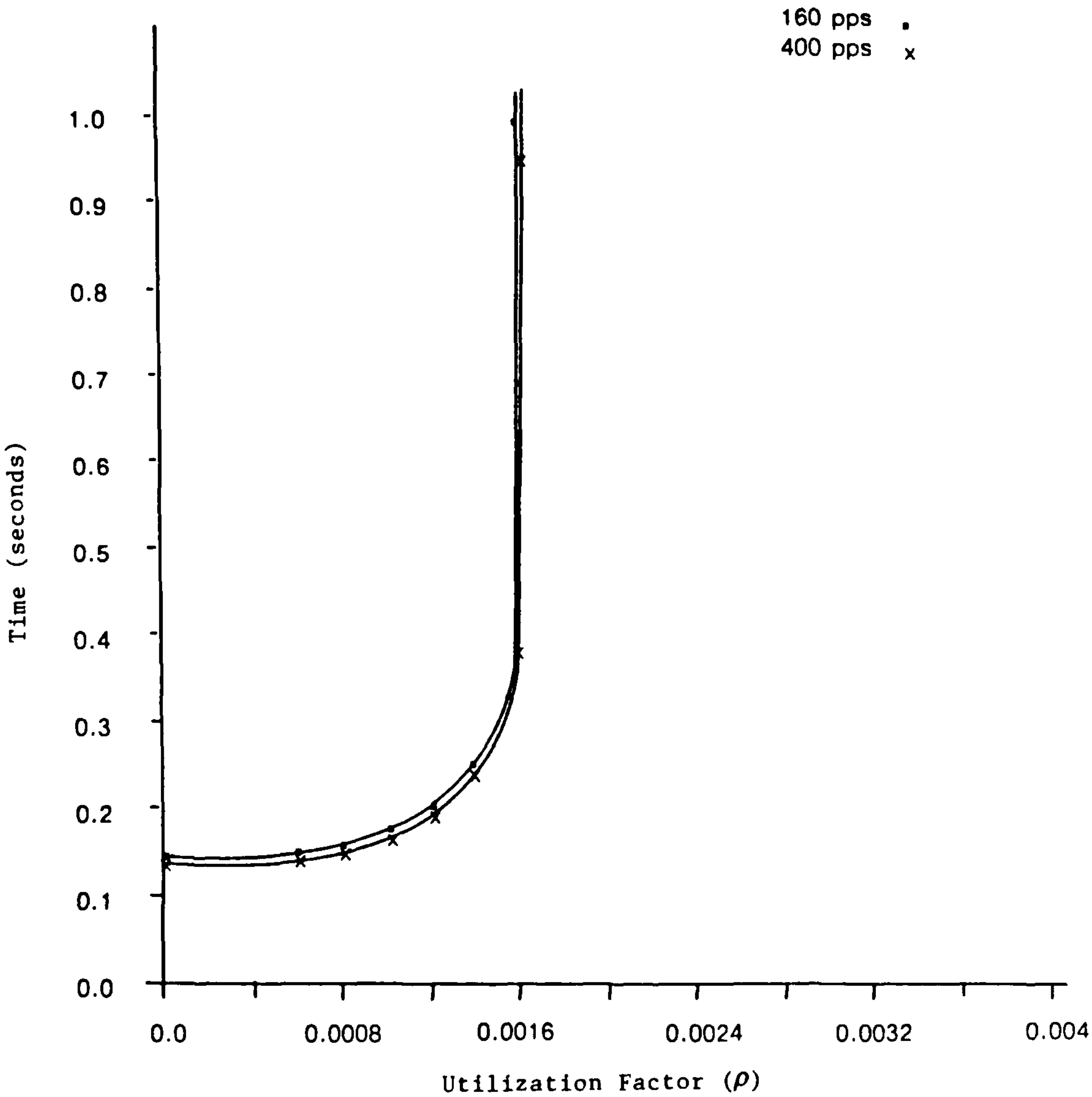


Figure 13. Average Delay with 9600 b/s ETS-PS Link

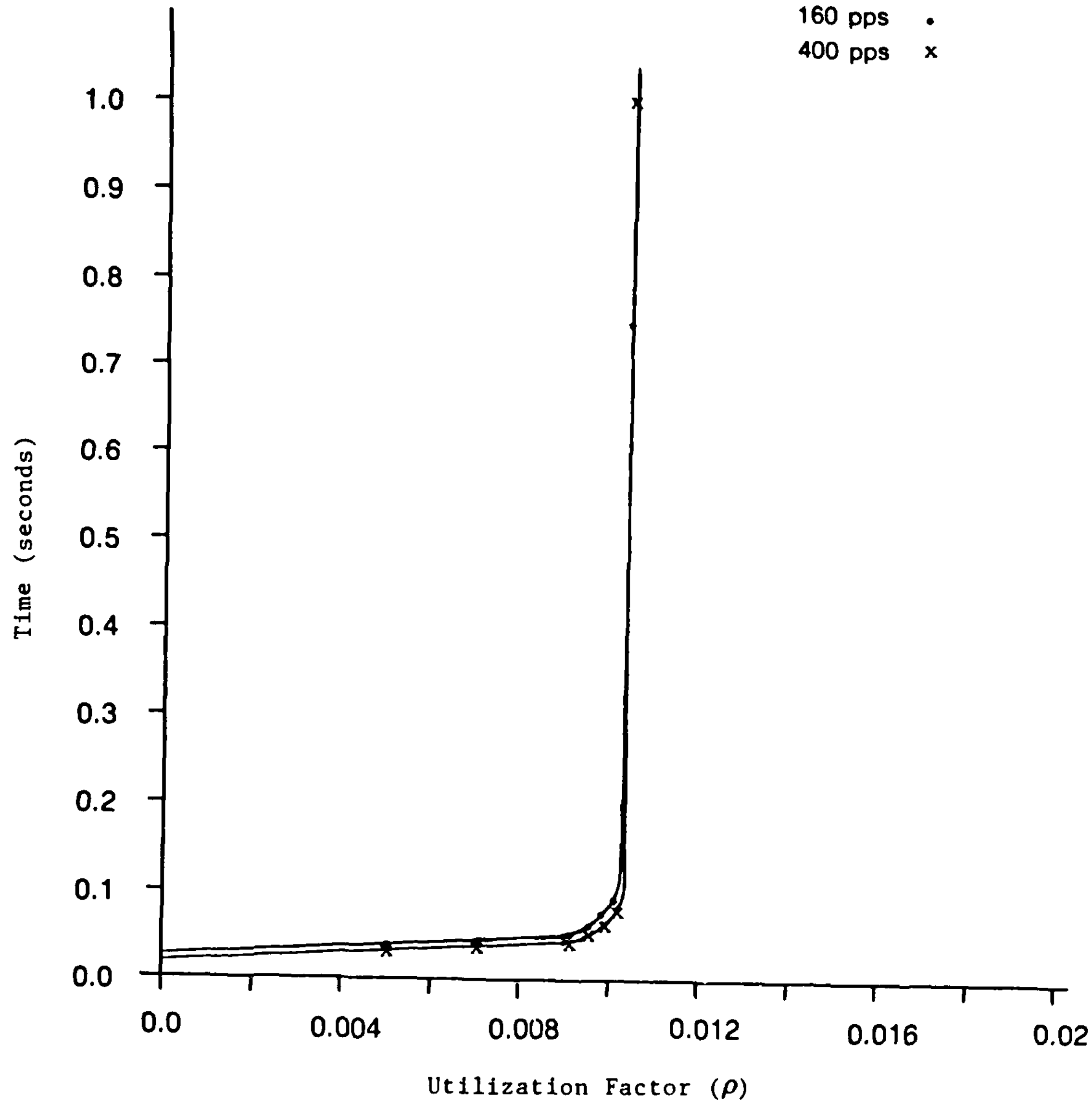


Figure 14. Average Delay with 64K b/s ETS-PS Link

The actual knee occurs near a utilization factor of 0.009. The average delay in this configuration can be expected to be less than 0.08 seconds.

In Run 1 and Run 2 it was shown that the throughput is limited to a degree by the ETS-PS link transmission rate. The higher the percentage of the terminal traffic that must traverse the ETS-PS link, the more significant that link becomes. It was also shown that the network delay is not significantly affected by the speed of the two packet switches.

Figure 14 also shows how channel transmission rate effects the average network delay. If the network traffic input is maintained at the intensity for 144K bits/second gateway channels but transmitted at 16K bits/second, the ISDN D channel rate, the delay curve is translated upward. Throughput is not effected by this change.

Errors. It would be expected that transmission errors would increase the average network delay. Figure 15 shows the effect on average network delay due to transmission errors on all links. The simulation was run with 160 packets/second and a 64K bits/second link. As the percentage of errors increase, the average network delay clearly increases. It can also be seen that as the error rate increases to and above 10%, that the retransmissions begin to saturate the network sooner then at the lower rates. A 10% error rate increases the average delay by

about 15%, a 25% error rate increases the delay by about 75%, and a 50% error rate increases the delay by about 200%. Note that the average network delay increase is greater than the percent of error retransmissions. This is due to the retransmission packets that must be sent after each NAK.

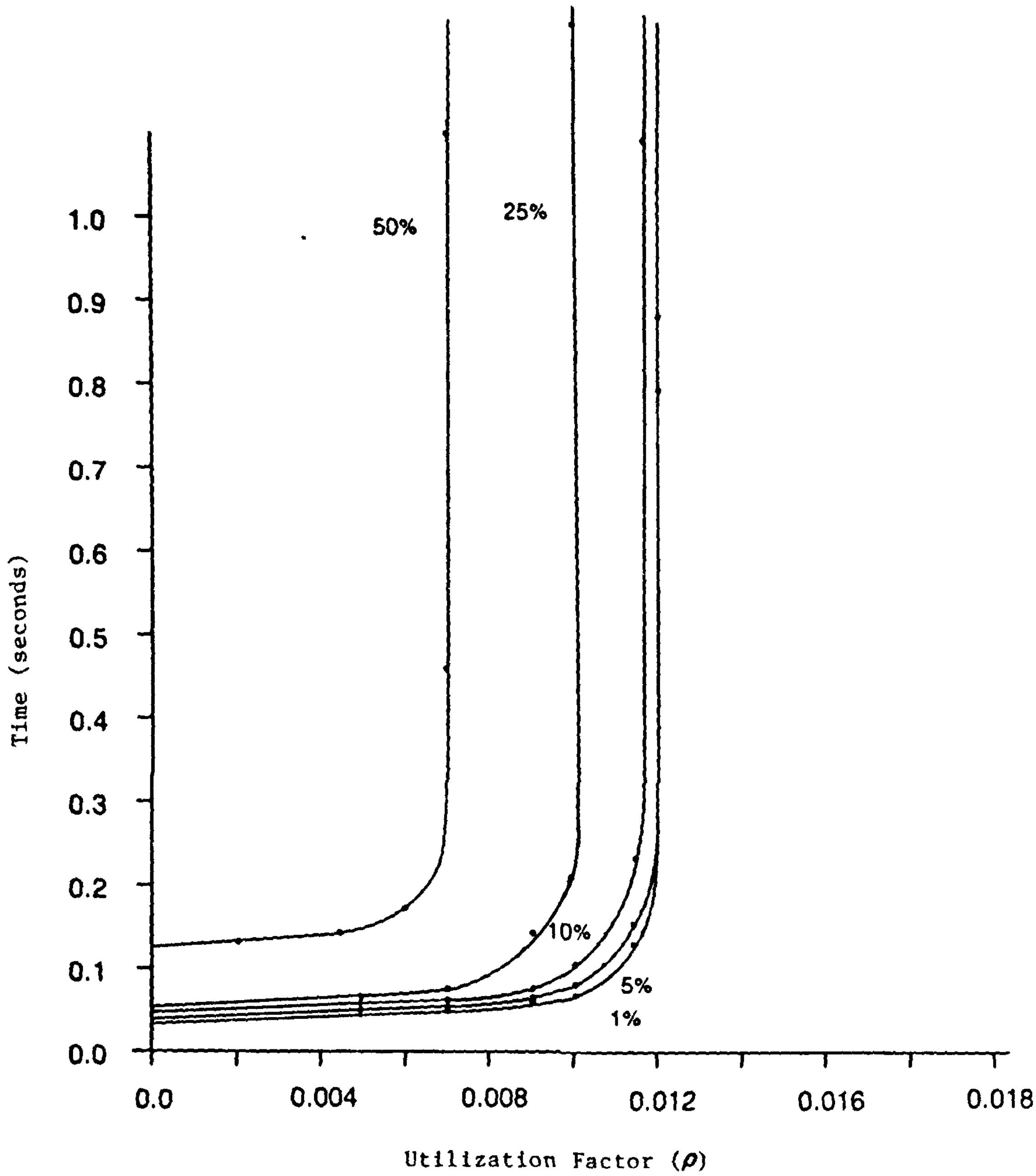


Figure 15. Average Delay Due To Transmission Errors

V. CONCLUSIONS AND RECOMMENDATIONS

Summary

The purpose of this thesis project was to develop a simulation model to represent and analyze the proposed ISDN oriented COMNET backbone of the Heidelberg information community. SLAM II was chosen as the simulation language because of its flexibility and expandability. A SLAM II model was validated against widely accepted analytical methods. The model was then expanded and modified to reflect the Heidelberg configuration. Several assumptions were made about unknown parameters and were based on published information about similar systems. Experiments were developed to exercise the model and examine the effects of changing several of the parameters that are critical to the performance of the network.

Conclusions

Phase I Network. The Phase I network consisted of the packet switch and only hosts connected with 64k bits/second transmission links. The resultant delay curves for the Phase I network show that throughput is limited by the packet switching rate. The average delay can be expected to be below 0.08 seconds while operating at a network packet arrival rate below the packet switching rate. The knee of the delay curve occurs at a utilization factor corresponding to the packet switch rate. It is also shown that even

though a C/300 packet switch processes 40% more packets per second than a C/30, the average network delay is not significantly effected.

Phase II Network. The Phase II network is the expansion of the Phase I network to the full system with terminal traffic connected to the packet switch through the ETS and the ETS-PS link. Consistent with the Phase I results, it was shown that an insignificant delay improvement was achieved when a C/300 packet switch was compared to a C/30. Equation 2 showed that delay is a function of channel capacity, packet size and the packet arrival rates in the ideal model. However, in the non-ideal network the node processing also contributes significantly to the average network delay.

The ETS-PS link was considered, in this model, to be part of the node processing overhead. It was shown that a system with the 9.6K bits/second link had an average delay 3.66 times greater than one with the 64K bits/second link. The transmission rate of that link was also a major limiting factor on throughput. The delay curves clearly showed a queue saturation point near the utilization factor corresponding to the bits/second rate of the ETP-PS link. Also, the more traffic traversing the ETS-PS link versus that remaining in the host side of the switch, the closer to the link rate will the throughput be limited. It should be pointed out that if the link transmission rate was increased

above that of the packet switch rate, then the packet switch rate becomes the limiting factor.

The maximum average network delay for the 9.6K bits/second ETS-PS link configuration not operating in saturation is 0.18 seconds. That same value for the 64K bits/second ETS-PS link configuration is 0.08 seconds.

The effects of transmission errors was also included. It was shown that error rates below 10% will not significantly degrade performance. However, at 10% and above, the number of retransmissions, in response to the NAKs, fills the queues faster causing network saturation. At a 50% error rate the increase in delay is 200% above the 0% delay curve.

A summary of the conclusions is:

Phase I Network:

a) The maximum average network delay can be expected to be 0.08 seconds.

b) The average network delay curve knee occurs at the packet switch rate, which is the throughput limiter.

c) A packet switch with higher throughput than a C/30 will not significantly decrease average network delay as long as the ETS-PS link rate is less than the C/30.

Phase II Network:

a) The maximum average network delay with the 9.6K bit/second ETS-PS link can be expected to be 0.18 seconds.

b) The maximum average network delay with the 64K bit/second ETS-PS link can be expected to be 0.08 seconds.

c) As long as the ETS-PS link bit rate is less than the C/30 packet switch bit rate, a faster packet switch will not significantly effect network delay or throughput.

d) The ETS-PS link is the throughput limiter as long as it is slower than the packet switch.

e) The greater the proportion of terminal traffic from the ETS, the closer to the ETS-PS bit rate is the throughput limited.

Transmission Error Rate:

a) Transmission error rates below 10% do not significantly effect performance.

b) The percentage increase of network delay is greater than the corresponding percentage increase of error rate due to packet retransmissions.

Recommendations

a) Packet Switch. Based on the results of the simulation runs, there appears to be consistent evidence that as long as the ETS-PS link transmission rate is below that of the C/30 packet switch, a faster switch is not justified. Therefore a C/30 that switches 750 bit packets at 160 packets/second will have sufficient capacity.

b) ETS-PS Link. The major limiting factor in the network simulation runs was the ETS-PS link. It is recommended that the 64K bits/second link be considered in place of the 9.6K bits/second link in order to maximize the throughput and keep the average delay to a minimum. If a

link capable of greater than 64K bits/second is chosen, then the packet switch rate may become the limiting factor and the switch capabilities should be reevaluated.

Further Research. The sponsor should use the model developed in this thesis as a basis upon which to expand toward a more complete ISDN system model. The following items should be done to further build upon the results obtained in this thesis for the COMNET backbone.

a) An assumption of this model was that all gateway traffic was equally distributed since there were no network input traffic figures available. As stated previously, this assumption should theoretically give worst case results for the average user. But, studies of the individual COMNETS should be done to determine more accurately how much traffic can be expected from each gateway in order to develop a model that more accurately represents the functioning real system.

b) After the traffic distribution is more accurately determined, SLAM II models of each COMNET can be built to verify the studies. The power of the simulation model is really needed in this case where expansion with unequal inputs may need to be modeled.

c) Separate backbone model input modules can then be constructed if any gateways significantly differ in the expected traffic input. It would be expected that most organizational COMNETs' input will be typically similar and

could be modeled closely with the techniques of this thesis. Possible exceptions could be expected from a personnel organization with a heavy inquiry rate.

d) If the ETS-PS bottleneck is resolved and an accurate traffic input is determined, then a particular channel may be found to be a bottleneck. Multiple channels may be represented since the simulation model represents channels as SLAM II resources. The number of channels can be specified in the resource statement of the code. As it now stands with a 9.6K bits/second ETS-PS link, concerns about a transmission link overload, for example the one to the personnel center at the Kilbern Barracks, is not an issue due to that ETS-PS bottleneck.

e) When the model of the physical network is matured and considered to be a confident representation of the real system, then various services that ISDN offers could be included in the model. The dynamic channel assignments in a simultaneous data/voice connection is one example.

These recommendations reflect the direction and sequence of further development that would have been taken in this thesis if time and resources would have allowed. Every effort has been made to make the model clear and expandable so that the sponsor can continue the effort through other theses or through in-house efforts.

Appendix 1: Variable and Attribute Descriptions

Variables.

XX(1) and XX(2). The interarrival rates are determined from $1/\gamma$. If gateway traffic is 80% of total network channel capacity and host traffic is 20%, then XX(1) and XX(2) are likewise proportioned.

XX(3), XX(4) and XX(5). XX(3) contains the mean value to be used for the exponentially distributed packet size. If an upper and lower limit to that packet size is used, then XX(4) contains the minimum and XX(5) contains the maximum.

XX(6). Holds the value of the ETS to packet switch (ETS-PS) link rate.

XX(7). The ACK/NAK packet size was determined from typical packet overhead of 64 to 256 bits (13:83). A value of 256 bits was considered to be a reasonable ACK/NAK packet size based on this source.

XX(8). The transmission error rate is used as the probability branch value in the channels and ETS-PS links.

XX(9) and XX(11). Source and destination calculations in the Switch Processing module use these values.

XX(10). Equivalent to the variable PSRATE. Contains the packet switching rate of the packet switch.

XX(12). Overhead delay of the ETS. Considered to be negligible in this thesis, but available for future use.

Attributes

Attribute 1. Contains the creation time of an entity representing a packet entering the network.

Attribute 2. Equivalent to variable DEST, the destination of the entity.

Attribute 3. Equivalent to variable SRC, the source node where the entity was created.

Attribute 4. Equivalent to variable LINK. Derived from SRC or DEST. Identifies the channel over which a packet is to be transmitted.

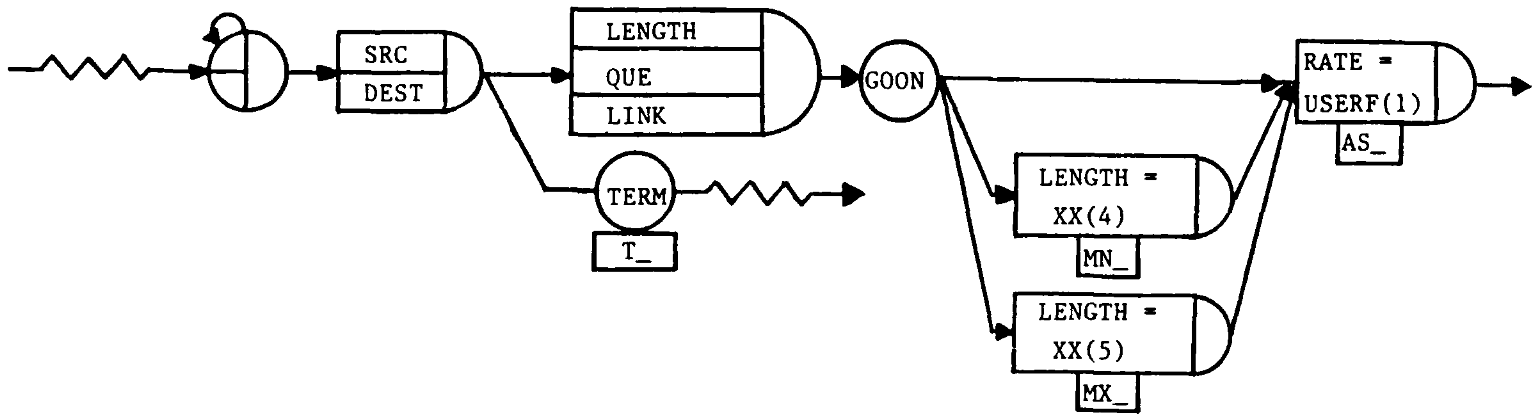
Attribute 5. Equivalent to variable QUE. Derived from SRC or DEST. Identifies the queue in which a packet waits for a free channel.

Attribute 6. Equivalent to variable LENGTH. Holds the exponentially determined packet length used in the transmission delay calculations in each transmission link.

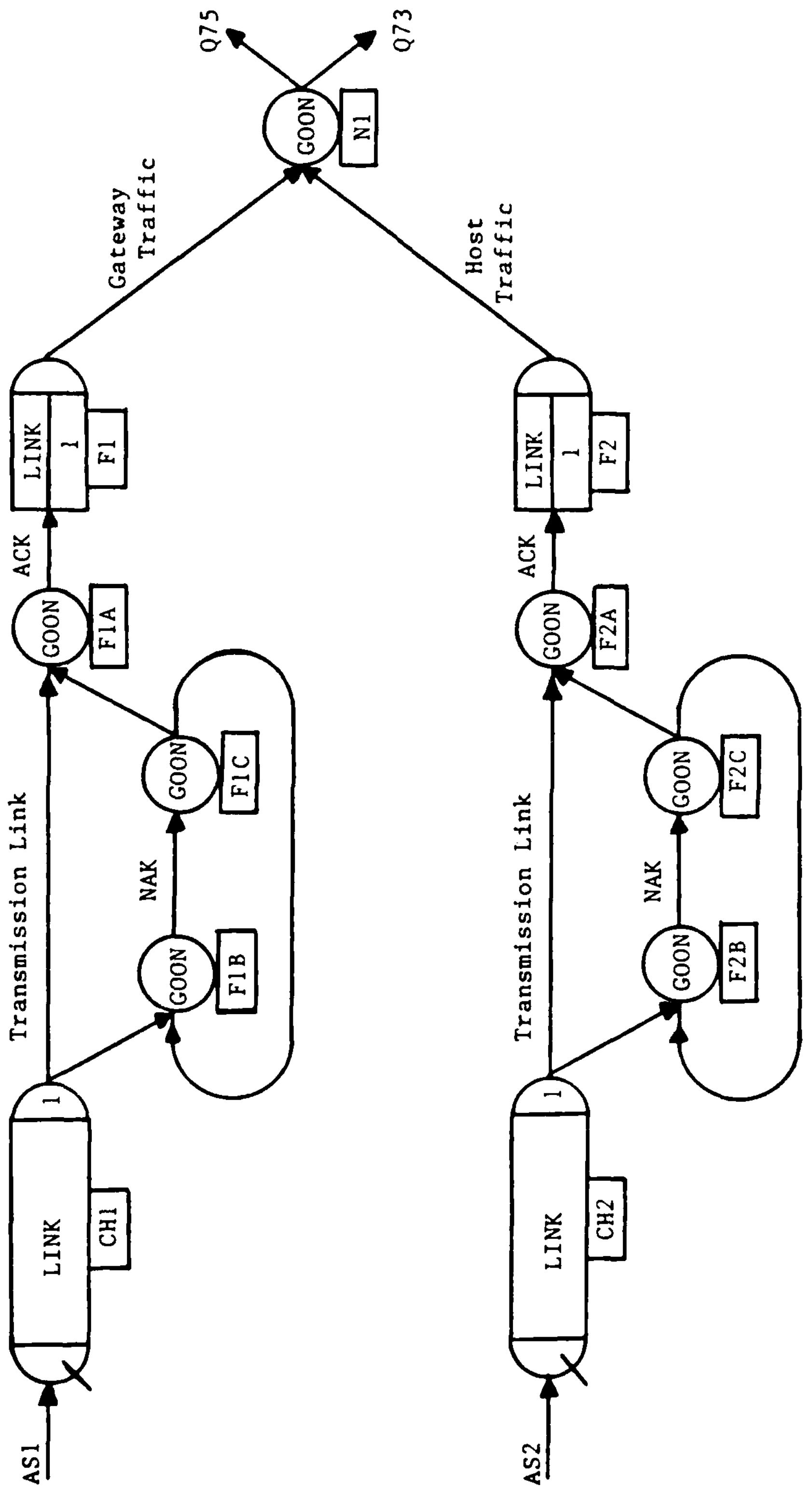
Attribute 7. Equivalent to variable RATE. Holds the transmission rate for a particular packet while waiting in a FIFO queue. It is determined in the user written function USERF(1).

Appendix 2: <u>Full Model SLAM II Network Diagrams</u>

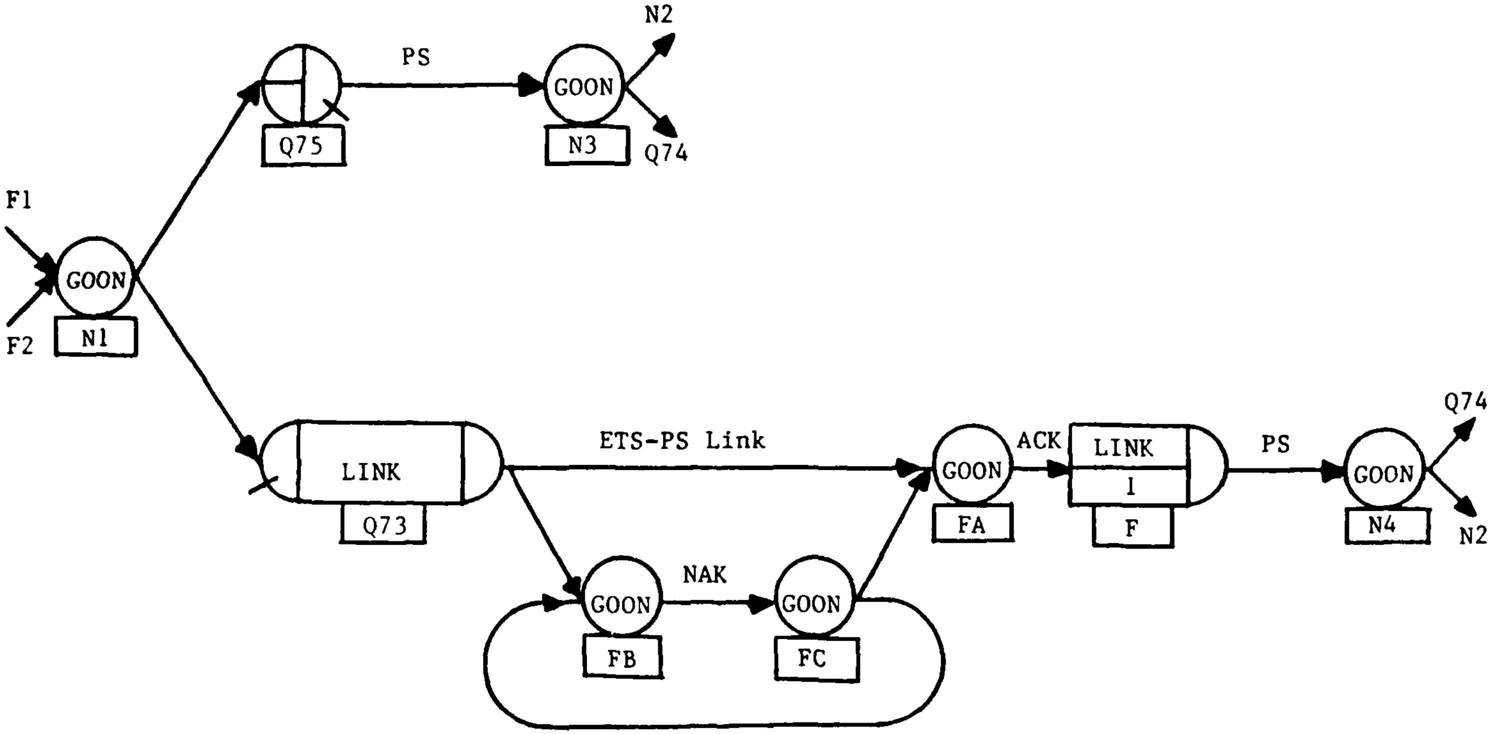
The following pages of Appendix 2 contains the SLAM II network diagrams the represent the full model of the Heidelberg backbone.



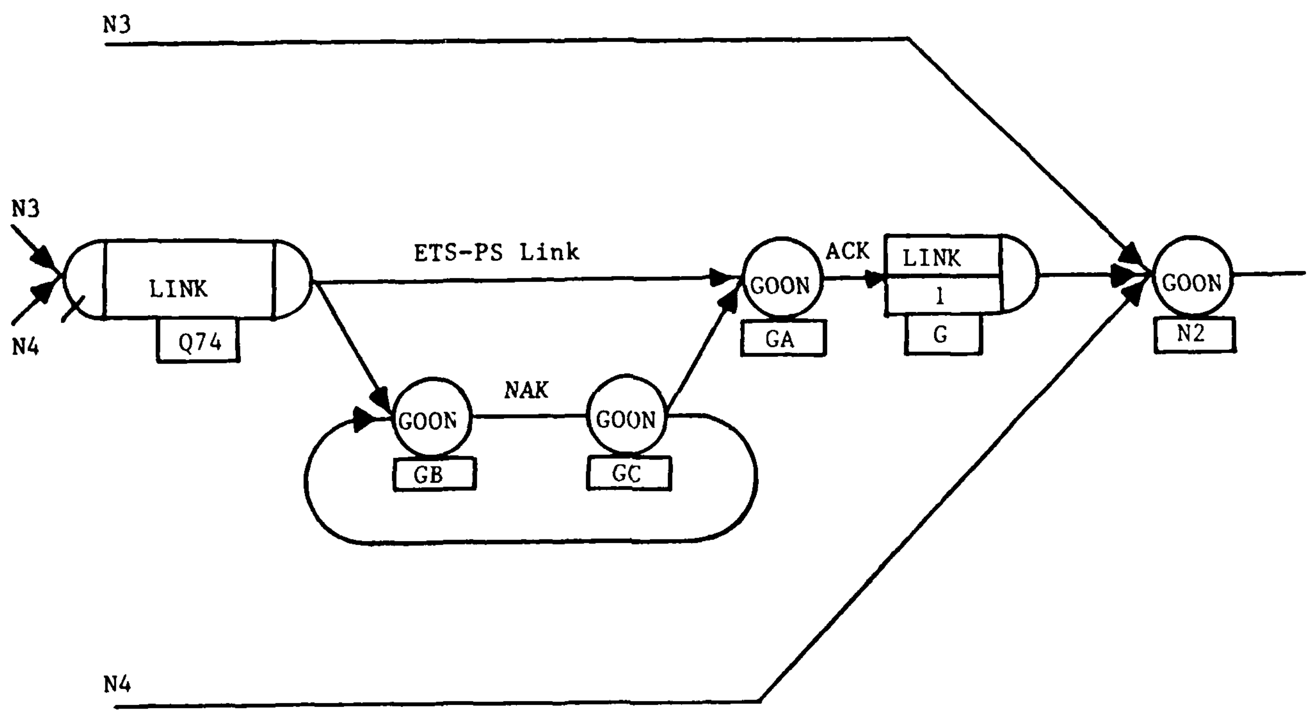
Input Module



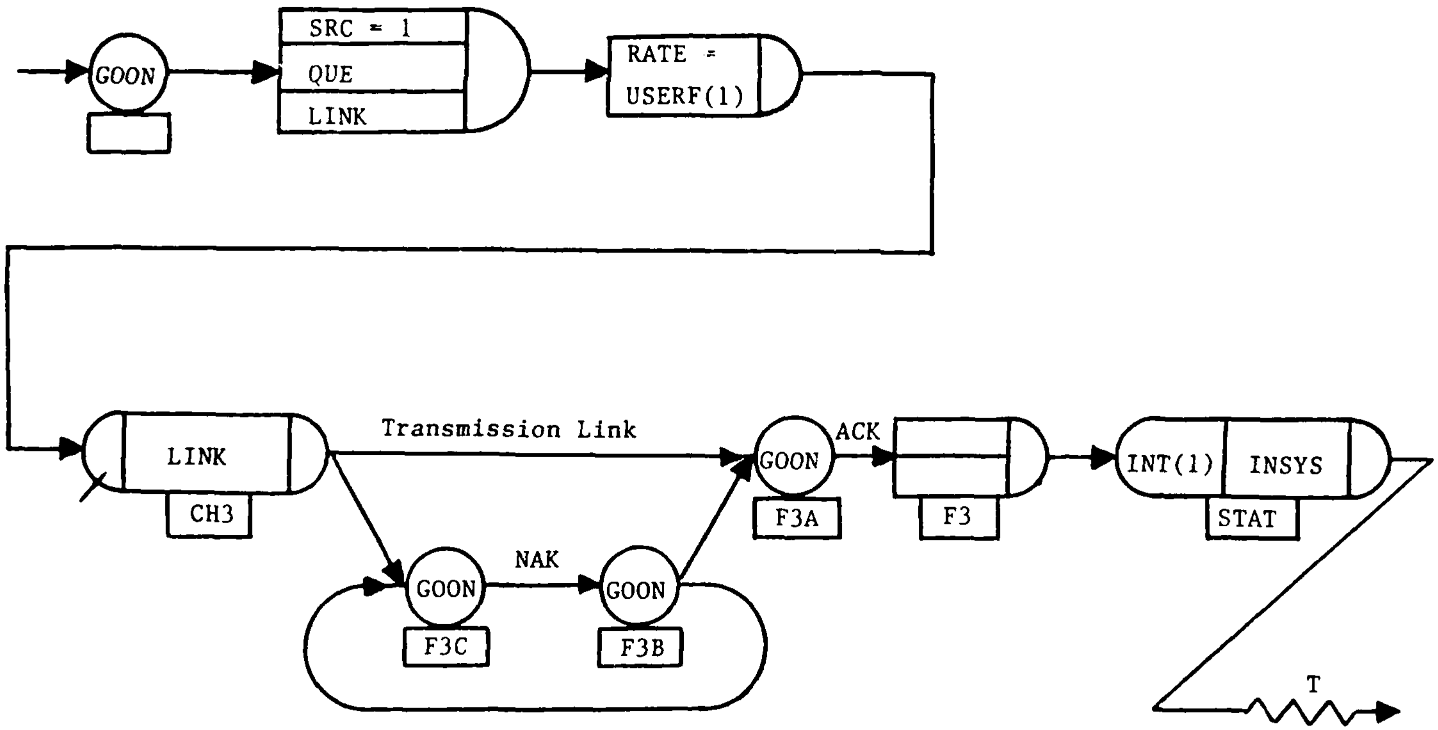
Transmission Module



Switch Processing Module



Switch Processing Module (continued)



Output Module

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VITA

Captain Lawrence J. Shrader was born on 6 July 1954 in Hanover, Pennsylvania. He graduated in 1972 from Delone Catholic High School, McSherrystown, Pennsylvania. He attended The Pennsylvania State University, from which he received the degree of Bachelor of Science in Electrical Engineering Technology in 1976. Upon graduation he was employed in plant engineering positions at Jones and Laughlin Steel Corporation until 1978, and Miller Brewing Company until 1981. He received a commission from the United States Air Force through the Officer Training School in 1982. He was assigned to the Command and Control Systems Office at Tinker Air Force Base, Oklahoma until entering the School of Engineering, Air Force Institute of Technology, in June 1986.

Permanent address: 173 Cemetery Street
Littlestown, Pennsylvania 17340

UNCLASSIFIED
SECURITY CLASSIFICATION OF THIS PAGE

REPORT DOCUMENTATION PAGE

Form Approved
OMB No. 0704-0188

1a. REPORT SECURITY CLASSIFICATION UNCLASSIFIED	1b. RESTRICTIVE MARKINGS
2a. SECURITY CLASSIFICATION AUTHORITY	3. DISTRIBUTION/AVAILABILITY OF REPORT Approved for public release; distribution unlimited.
2b. DECLASSIFICATION / DOWNGRADING SCHEDULE	
4. PERFORMING ORGANIZATION REPORT NUMBER(S) AFIT/GCS/ENG/87D-25	5. MONITORING ORGANIZATION REPORT NUMBER(S)
6a. NAME OF PERFORMING ORGANIZATION School of Engineering 6b. OFFICE SYMBOL (If applicable) AFIT/ENG	7a. NAME OF MONITORING ORGANIZATION
6c. ADDRESS (City, State, and ZIP Code) Air Force Institute of Technology Wright-Patterson AFB OH 45433-6583	7b. ADDRESS (City, State, and ZIP Code)
8a. NAME OF FUNDING / SPONSORING ORGANIZATION 8b. OFFICE SYMBOL (If applicable)	9. PROCUREMENT INSTRUMENT IDENTIFICATION NUMBER
8c. ADDRESS (City, State, and ZIP Code)	10. SOURCE OF FUNDING NUMBERS PROGRAM ELEMENT NO. / PROJECT NO. / TASK NO. / WORK UNIT ACCESSION NO.

11. TITLE (Include Security Classification)
See Box 19

12. PERSONAL AUTHOR(S)
Lawrence J. Shrader, B.S., Capt, USAF

13a. TYPE OF REPORT	13b. TIME COVERED	14. DATE OF REPORT (Year, Month, Day)	15. PAGE COUNT
MS Thesis	FROM ______ TO ______	1987 December	83

16. SUPPLEMENTARY NOTATION

17. COSATI CODES			18. SUBJECT TERMS (Continue on reverse if necessary and identify by block number)
FIELD	GROUP	SUB-GROUP	Local Area Network, ISDN, Simulation, SLAM II

19. ABSTRACT (Continue on reverse if necessary and identify by block number)

Title: SIMULATION AND ANALYSIS OF A LOCAL AREA NETWORK INTERCONNECTION BACKBONE FOR HQ, U.S. ARMY EUROPE AND THE SEVENTH ARMY

Thesis Chairman: Albert B. Garcia, Lieutenant Colonel, USA
Assistant Professor of Electrical Engineering

24 Feb 88

20. DISTRIBUTION / AVAILABILITY OF ABSTRACT ☒ UNCLASSIFIED/UNLIMITED ☐ SAME AS RPT ☐ DTIC USERS	21. ABSTRACT SECURITY CLASSIFICATION UNCLASSIFIED
22a. NAME OF RESPONSIBLE INDIVIDUAL Albert B. Garcia, Lt Col, USA	22b. TELEPHONE (Include Area Code) 513-255-3576 22c. OFFICE SYMBOL AFIT/ENG

DD Form 1473, JUN 86 Previous editions are obsolete

SECURITY CLASSIFICATION OF THIS PAGE
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This thesis project develops a simulation model of a proposed backbone that will interconnect approximately 16 existing local area networks of the information community at HQ, U. S. Army Europe and the Seventh Army in Heidelberg, Germany. The objective is to develop a computer simulation that determines throughput and delay characteristics of the backbone. Potential configuration problems that may limit the full utility of the system are also identified from results of the simulation.

An existing telephone switch at the Campbell Barracks in Heidelberg will be upgraded and interfaced, by way of a 9.6K bits/second link, to a packet switch to provide an Integrated Services Digital Network (ISDN) environment for the integration of data and voice in one switching system. The computer simulation model is developed to analyze the data portion of the proposed system only.

The simulation is implemented using Simulation Language for Alternative Modeling (SLAM II). The network orientation of SLAM II is used to represent the system with a user written FORTRAN function provided to determine the trans-mission link rates.

A valid model is developed from a widely accepted analytical analysis of a star network similar to the proposed system at Heidelberg. The final validated model is then exercised with two different packet switch rates and two different link speeds of the telephone switch-to-packet switch interface.

The simulation results show that with the originally proposed configuration, the packet switching rate was not a significant factor in throughput and delay measures. The proposed 9600 bits/second interface between the packet switch and existing telephone switch is the limiting factor of the two performance measures.

Among the recommendations is the suggestion that an interface with a higher bit rate be considered since this was the major limitation of the proposed system.

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